Documentation

HiPath 4000 V5

Feature Description

A31003-H3150-F100-1-7618

Communication for the open minded

Siemens Enterprise Communications
www.siemens.com/open

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Reference No.: A31003-H3150-F100-1-7618

HiPath 4000 V5 - Feature Description - Contents

1 Preliminary Remarks

1.1 Introduction

The descriptions of the features applies to the Federal Republic of Germany and to the countries listed in Chapter 10, “Country-Specific Features”. Please regard that certain features may be inconsistent with individual country specific regulations.

For additional information and descriptions, please refer to the service manuals.

Type of telephones

- analog telephones
- **Digital system telephones:**
  system telephones, which attached to an Up0E-Interface are controlled by the company-specific CorNet-protocol.
- **IP telephones:**
  IP telephones which are controlled by the company-specific CorNet protocol HiPath Feature Access.
- **SIP telephones:**
  IP telephones which are controlled by the SIP protocol.

IMPORTANT: This description is not to be considered as a commitment that all functions contained are available on the general system release date.

1.2 Functions that have been added or adapted:

1. Addition: chapter 2.2.6
   Signaling and payload separation
2. Modification: chapter 2.3.3
   SIP- subscriber
   Support of new features
3. Modification: chapter 2.3.4 (formerly IP-trunking mode)
   SiP-Q-trunking
   SIP-Q-trunking replaces CorNet IP
   Support of new features

4. Modification: chapter 2.3.5 (formerly SIP-trunking mode)
   Nativ SIP-trunking
   Support of new features
5. Modification: chapter 2.3.7
   Fax/Modem/DTMF
   Modification relating to T38 and DTMF
6. Modification: chapter 2.3.10
   Large enterprise gatekeeper
   Registration of remote HG 3500 at the HiPath 4000 LEGK will not be supported anymore
7. Addition: chapter 2.4
   Softgate
8. Addition: chapter 4.14
   Multi-Level-Precedence and Preemption (MLPP)
9. Addition: chapter 4.15
   CLI- modification
10. Modification:chapter 5.18
    HiPath relocate
    Emergency call from a telephone which is logged off
11. Modification: chapter 5.31
    Call log: call logging in the case of call transfer

1.3 Additional modifications

1. Deletion off all descriptions (e.g. hardware platform 600 ECX, 80CMX-DSC, VCM7, TMAU, TMBLN, TMEMW...), that are no longer supported by HiPath 4000 V5.
2. Modification according the new interface DIUT2.
3. Deletion of HiPath 5000 references because interworking to HiPath 5000 are no longer supported.

2 General

2.1 Overview

The innovative Real Time IP System HiPath 4000 enables the connection and use of TDM and IP based infrastructures. Open interfaces and standard protocols are the platforms for the integration of communication and data applications. A wide variety of multimedial communication forms are supported.

HiPath 4000 is ideal for mid-sized and huge companies, that attach great importance to security, flexibility and services. The system offers a fast and easy access to an unified voice- and datakomunication for everybody. A groundbreaking solution that optimizes your business processes and provides for a higher efficiency and productivity.

Improved for business

With HiPath 4000 communication takes the center stage of the business processes.

- A seamless network with a uniform utilization
- Smooth embedding of voice communication in your IT-network
- Instantly significant improvement of the workflow
- High reliability
- Broad compatibility with many different applications
- High return on investment, low cost of ownership and especially smooth implementation

A reliable, flexible and inexpensive end-to-end-solution

Among the benefits are:

- Voice- and data transmission in real-time
- High-class, well equipped terminals that are easy to use
- Support of SIP, therefore problem-free migration, easy integration and reliable protection of your investment
- Maximum protection against hackers, viruses and system failures
- Outstanding flexibility and scalable for up to 100.000 users
- Low administration-, installation- and integration costs
- Supports future features and adapts to changing demands

HiPath 4000 also offers company network providers additional features which allow marketing of PABX functions to selected companies (Business networks).

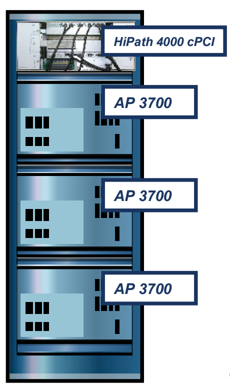
2.1.1 HiPath 4000 (cPCI) Communication Server

The hardware-concept of the DSCXL system consists of a cPCI based 19" shelf for the central switching units and the existing Flex-Shelves for the peripheral interface boards.

2.1.1.1 Housing of DSCXL

The HiPath 4000 system is provided with two different housings:

- 19"-Compact PCI sub-rack as mechanical core of the DSCXL system with air cooling



complete 19“ rack system

Figure 1 19" model

- Stand-Alone-Housing (when 19" rack is not available)

  The cPCI shelf is integrated into the CCDAX shelf with a mounting kit

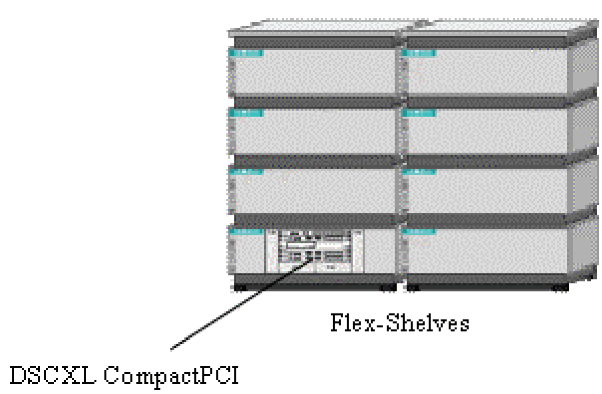


Figure 2 Stand-alone-housing

All system configurations can be run with one PSU only. With a second PSU redundancy is provided. Hot-swapping is supported.

A Battery Manager Interface for -48V Battery Mode ist supported

One DSCXL cPCI shelf supports:

– up to 15 directly connected peripheral shelves

– the system configurations Mono and Duplex

The possible Hardware configurations of the system are shown in the next figures:

- **Mono**

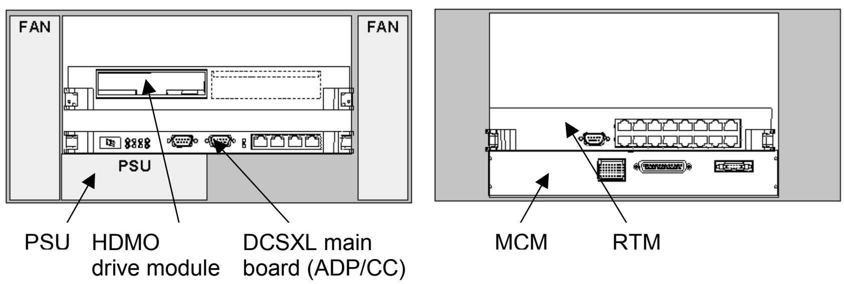


Figure 3 Mono configuration

- **Duplex**

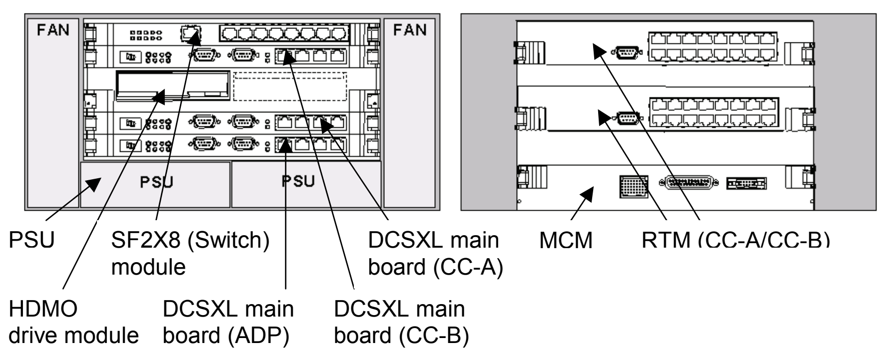


Figure 4 Duplex configuration

2.1.1.2 Hardware concept

The DSCXL systems consist of a set of 5 different modules:

Main System Board DSCXL

The DSCXL Main Board is based on the Intel Pentium III processor.

DSCXL Rear Transition Module RTM

The RTM module supports the following functions:

- Fit in cPCI system
- Support for 15 LTU-Shelves
- Replacing current LTU cable by standard CAT5 cable (RJ45-8wire)
- Interface to DSCXL via LAN-Interface

The RTM module integrates the SIU function: An extended number of 12 DTMF receivers and transmitters are realized.

SF2X8 LAN Switch Module

The SF2X8 module offers the 10/100 Mbit LAN switch for the LAN ports of the cPCI backplane and a second independent 10/100 Mbit LAN switch for customer use. Each switch supports 48 LAN connectors.

DSCXL module

The Hard Disk module is based on IDE drives (operated in DMA mode). Access to the each drive is possible via the PCI bus of the cPCI system through the PCI to IDE bridge. The module supports basic hot swap functionality.

Management and Control Module MCM

The MCM is a module, which supports several interfaces:

- the fan control unit,
- the ALUM (Amtsleitungsumschaltung),
- the ALIN (Alarm Interface) and
- the Frontreference Interface (Interface to the external clockbox)

The module supports hot swap.

The modules described above are integrated in a 19"-subrack (or a optional stand-alone-housing) and communicate via backplane-integrated LAN.

Only the HDMO-module is accessed by the ADP via PCI-Bus. The central control (CC-A/B) has no access to the cPCI-bus.

In a DSCXL system a standard LAN cable (shielded CAT5) is used to connect the today's peripheral shelf to the switching unit.

Overview cPCI

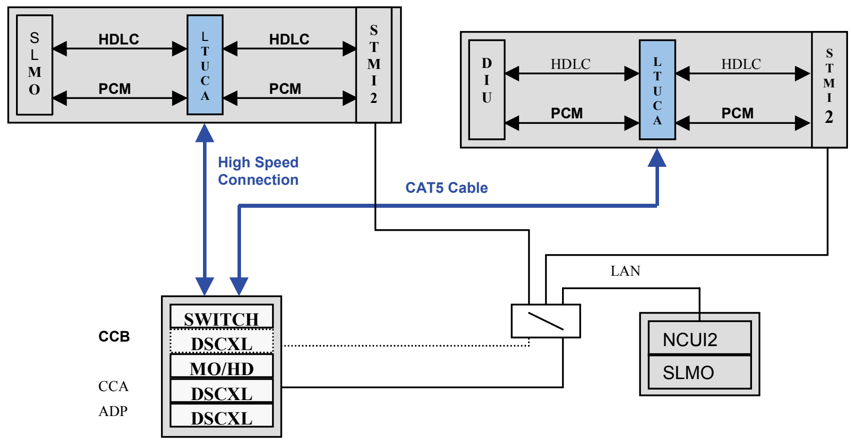


Figure 5 Overview cPCI

2.1.1.3 Line Trunk Unit Control Advanced (LTUCA)

The LTUCA (Line Trunk Unit Control advanced) is the interface between central control and peripheral parts in the HiPath 4000 architecture. The LTUCA card only supports the architecture of HiPath 4000 as of V2.0, is not backward compatible. LTUCA supports the following backplane versions:

– LTUW

– L80XF

– L80XW

– AP 3700 (with 13 slots)

The central unit (DSCXL/RTM) and the peripheral shelves are connected via a high-speed connection with a standard LAN cable (CAT 5).

2.1.2 19" Access Point AP 3700

The access point AP 3700 is a 19" shelf for HiPath 4000 systems in the following variants:

– AP 3700 IP, 9-slot box with additional slot cPCI cassette as a IPDA shelf with emergency option

– AP 3700, 13-slot box as a shelf within HiPath 4000 stand alone system

Features

The AP 3700 IP and the AP are almost identical in their mechanical structure. The sheet metal frame is divided by the mid-plane into module side and patch panel-/external cable side. The module side itself consists of two logical units:

– card cage (module pitch 30mm)

– power supply area (PSU pitch 90mm)

An additional space is reserved in the AP 3700 IP for:

– cPCI module (6Ux48TE) (AP-CC for emergency option)

The patch panel-/external cable side accommodates the corresponding patch panels (pitch 30mm) or cables for the peripheral boards.

The AP3700 IP and AP 3700 are both designed for free convection cooling. The environment temperature must be less or equal than 45° C. The optional Compact PCI module (in AP 3700 IP) needs forced air-cooling. In case of stacking systems (especially in 19" racks) each configuration has to be approved by compliance.

Variants

The housing is designed for stand-alone use or 19" rack integration.

Stand-Alone system

The housing is designed to be used as stand-alone system. To protect the system interfaces and for industrial design reasons, an optional front and rear cover can be installed. To meet compliance requirements (e.g. earthquake), for stacking additional mounting kits may be necessary.

19"-rack integration

With optional mounting brackets, the housing can be integrated into a standard 19" rack. The position within the rack has to be approved by compliance (thermal aspects).

AP 3700 IP

Front view (module side):

– 1x central board HG 3575

– up to 9 peripheral boards

– up to 3 power supplies

– 1x optional cPCI cassette (AP Emergency) with own power supply

Rear view (patch panel side):

– up to 9 patch panels (8-, 16- and 24 port RJ-45 version, CHAMP version, HiPath 3750 MDF)

Rear view without patch panels:

(if no integrated patch panels are used)

– Cable connectors SIVAPAC to external patch panels or MDF

– Power input panel (DC top / AC bottom)

AP 3700

Front view (module side):

– 1x central board LTUCA

– up to 13 peripheral boards

– up to 4 power supplies

Rear view (patch panel side):

– up to 13 patch panels (8-, 16- and 24 port RJ-45 version, CHAMP version, HiPath 3750 MDF)

Rear view without patch panels:

(if no integrated patch panels are used)

– Cable connectors SIVAPAC to external patch panels or MDF

– Power input panel (DC top / AC bottom)

cPCI module (only in AP 3700 IP)

The CompactPCI cassette is a sub-system consisting of housing, backplane, processor
modules, mass storage module, power supply and fans.
It can be integrated into the AP3700 IP (for emergency option = AP Emergency) and is connected via LAN cable (CAT5) to the central board of the AP 3700 IP.

Remarks:

– MCM board (fan control, 48V power box failure signalling ALIN, ALUM,...) and RTM board are not supported in AP 3700 IP

– Redundant Power supply of the cPCI cassette is not supported

– The cPCI cassette needs a fan driven forced cooling

Power supply concept (LUNA2)

The LUNA2 power supply unit:

– is air cooled by free convection only by a max. DC output power of 172 W,

– operates in free space at 55° C at a nominal voltage of 240 VAC and 45°C ambient temperature, at a nominal voltage of 115 VAC,

– provides active power sharing of more than 2 up to max. 4 power supplies going in parallel. LUNA2 supply can be changed between operating in case of redundancy and serviceability, if the frontside switch is positioned to "OFF",

– has all connectors including its mains connection on the backplane side,

– has a bi-directional DC IN/OUT on its backplane interface, which allow to feed LUNA2 only from battery in mode 1 or to work as a battery charger output in mode 2,

– has a switch for selection of mode 1 or 2 which is visible on front of supply,

– has a Power fail signaling interface for HiPath 4000 (ALIN)

– NGA Signal: Signals mains power fail, low active and open drain,

– PFL Signal: Signals DC output voltage fail on one or more DC output rails, low active and open drain.

IMPORTANT: Since ALIN Interface is not supported in AP3700 and AP 3700 IP, an NGA-Signal coming from a Power Box UACD or UDCD is not supported. The NGA signal is used for functions related to power fail / battery backup mode signaling.

Redundancy of power supply

The power supply supports n+1 redundancy (without battery charging).

– AP 3700 IP:
2 LUNA2 support the AP 3700 IP, the 3.rd LUNA2 is for redundancy if 1 LUNA2 fails

– AP 3700:
3 LUNA2 support the AP 3700, the 4.th LUNA2 is for redundancy if 1 LUNA2 fails.

Ringing Generators

The Ringing generators RGMOD (Siemens), RGMOD (Magnetek) and RGE are necessary to provide ringing for SLMA24 analog subscribers. They are not necessary for SLMAE / SLMAR-users

Battery Boxes HiPath 4000

The use of UACD/UDCD HiPath 4000 Power Box is possible with AP 3700 (IP).

Remarks and Restrictions

General instructions for support in AP3700 (IP):

- Only boards with SIPAC-connector (Preferred)
- No conversion of old boards (Status 37 or higher) from AP3300
- Boards with up to 30mm height max.
- Limited amount of boards with extensive power consumption, e.g. SLC24
- Boards without front-connectors only (Preferred)
- No central board

Please see service manual and/or TI for an actual list with supported HiPath 4000 peripheral boards.

The cPCI cassette in AP 3700 IP is a reduced variant compared with the central unit module used in the HiPath 4000 platform - without the modules MCM and without redundant power supply.

The power supply of AP 3700 (IP) is possible only with LUNA2.

2.1.3 AP Emergency Concept

The solution covers IP network failures only. With "AP Emergency" a solution for the AP3700-9 is provided which covers the failure of the central HiPath 4000 host system.

AP Emergency means amongst others:

- The "AP Emergency" Solution can be used in addition to the already existing "survivability" features.
- With "AP Emergency" an additional (de)central control system is implemented that controls IP Access Points in emergency situations (CC-AP)
- A mechanism turns over the control from the host processor to the CC-AP and vice versa
- The switch over criteria can be configured independently for each IPDA AP /CC-AP. The switch over process can be done automatically. Besides the emergency features can be activated or deactivated manually.
- 1:1 AP Emergency is supported: In this case every AP 3700-9 of a system is equipped with its own CC-AP.
- It is possible, that a CC-AP controls more than 40 access points.
- For every standard subscriber (analogue, digital, attendant console) it is possible to define an alternate destination. This alternate routing can be activated, whenever the system control detects an unavailable device.

– A mechanism is implemented to "clone" the host database and install it on the CC-AP

2.1.3.1 Architecture and Feature Description

With "AP Emergency" additional control processors of IPDA access points (CC-AP) can be used to take over the control of a configured amount of access points when these have lost the connection to the host control processors (CC-A, CC-B).

Every CC-AP has a clone copy of the central database. Replication is done by extending the already existing backup/restore concept. Cloning is possible without a HiPath 4000 Manager available.
The important issue is the signaling or monitorage of the connectivity between the HG 3575 and an access point with it's central control.

The access points are located in different areas. CC-A / CC-B controls every access point
in the system.

All three access points (AP 17 - AP 19) are prepared for control by the CC-AP in AP 17.

– AP 20 and AP 21 have the CC-AP in AP 20 for emergency operation.

– AP 40 and AP 41 have the CC-AP in AP 40 for emergency operation.

The access points AP 20 / AP 21 and AP 40 / AP 41 are likewise equipped with an CC-AP.

Figure 6 shows the signaling connectivity in normal operation.

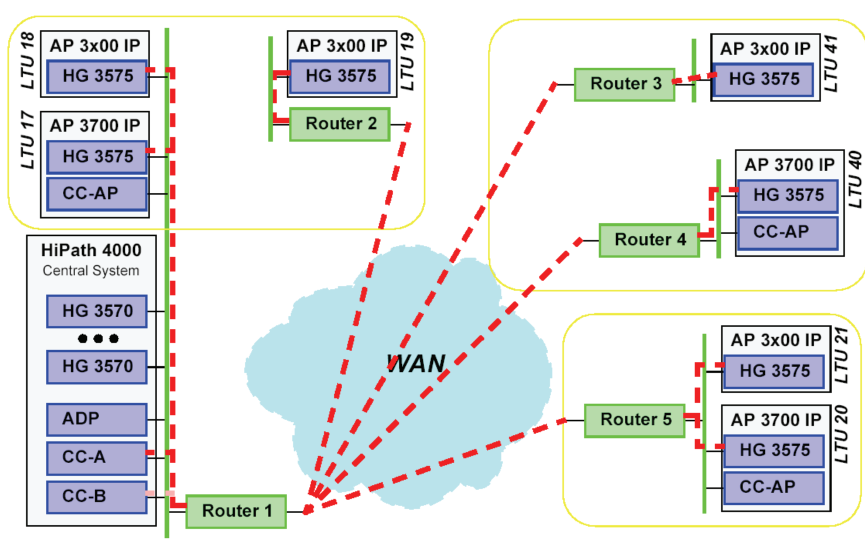


Figure 6 Signaling Flow in Normal Operation

In case of a host system outage, the CC-APs take over control. The single system temporarily breaks down into 3 separate systems.

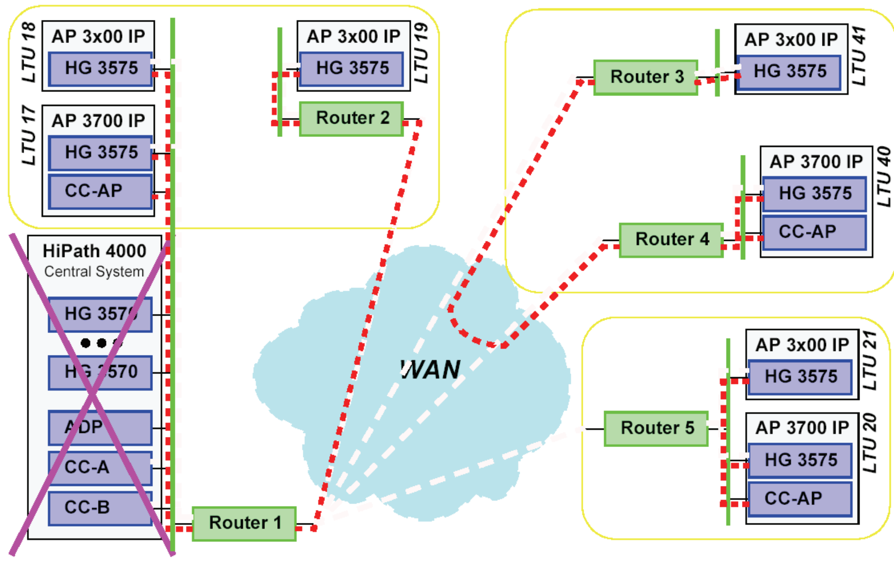


Figure 7 Signaling Flow at Host Outage

This is not the only scenario where the AP Emergency feature becomes effective. In scenarios without the survivability features AP Emergency also handles various situations of e.g. WAN

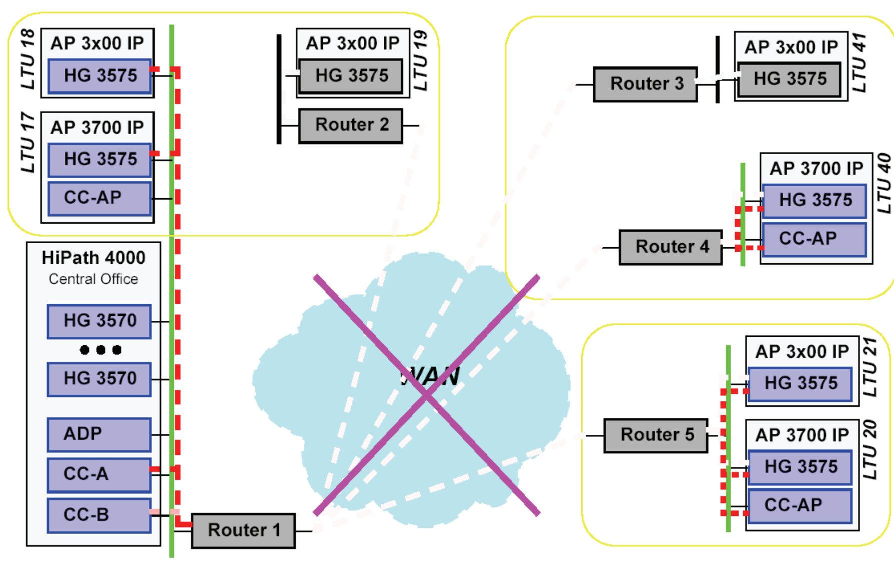


Figure 8 Signaling Flow at a total WAN Outage

2.1.3.2 Solution Concept

The feature "AP Emergency" is based on a set of functional blocks.
These are:

- Alternate Routing on Error
- Detection of the Emergency Situation and Switch Over
- DSCXL in AP3700-9 (CC-AP)
- Administration and Serviceability (A&S)
- Administration of the "AP Emergency" feature with HiPath Assistant and HiPath 4000 Manager

An algorithm is used to find out whether an access point has to be controlled by the host or by the associated CC-AP upon partial WAN/LAN failures. For this decision, configurable rules are provided.

The subscriber notices the switch over to and back from the AP Emergency mode, as calls are disrupted and the system will be unavailable for about 60 seconds. During startup of the AP Emergency mode the phones are out of service. To ensure voice connectivity, calls to subscribers of the system, which are outside of the scope of a CC-AP in AP Emergency mode, are routed via PSTN (feature *"Alternate Routing on Error")*.

The following items must be administered:

- IP addresses of the CC-APs
- Association of the AP to the controlling CC-AP
- Configuration rules for switch over to AP emergency operation
- Configuration rules for switch back to normal operation
- Time schedule for automatic database cloning

The Collection of call data records is done in the host system and in every CC-AP, thus no CDR information is lost. The HiPath 4000 Manager is necessary to collect and merge the CDR data from the CC-APs and the host.

Definition of Emergency Groups

Based on the idea that an AP shelf with the AP emergency feature is assigned directly to an CC-AP, this approach puts all assigned AP shelves of an CC-AP into a single AP group. But with only one AP group it would be difficult to define useful switch-over criteria that are suitable for all possible scenarios (from systems with a single CC-AP to systems with one CC-AP for each AP shelf).

Derived from it access points can be summarized (Emergency Groups), whereby per group only one CC-AP is installed.

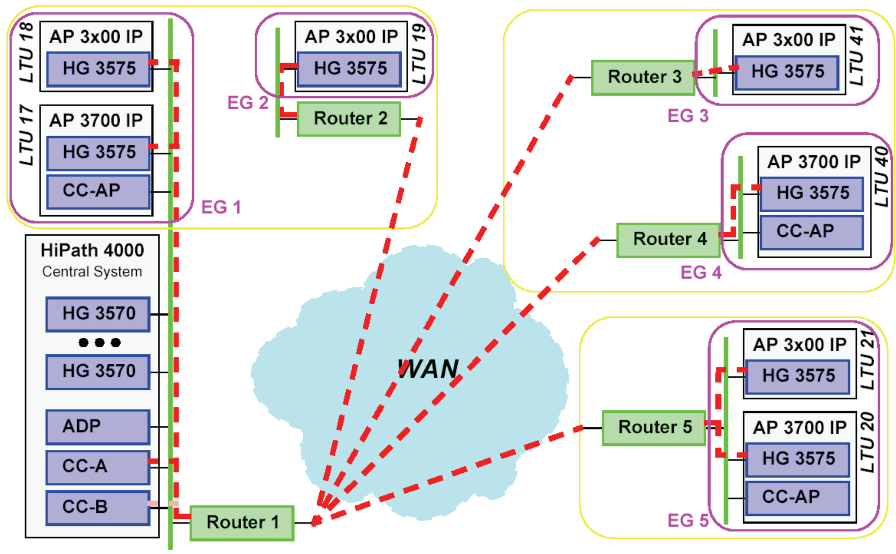


Figure 9 Definition of Emergency Groups

Switch over in emergency situations

If a HG 3575 detects problems with the signaling connection from the host system, it reports that fact to the associated CC-AP. The CC-AP decides according to preconfigured rules whether to take over control of the AP or not. Emergency mode and control by CC-AP is an escalation step after Signaling Survivability fails (if Signaling Survivability is configured).

Every AP has a weight - when the total weight of all associated APs that lost host connection is above a predefined threshold, the CC-AP takes over the control. The APs can be configured to be taken over individually by CC-AP.

Switch back in normal operation

If a HG 3575 detects that the connection to the host system is up again, it reports that fact to the associated CC-AP. The CC-AP decides according to preconfigured rules whether to give back control of the AP or not. The Switch back can also be instructed by the system administrator

To take over control of an AP, the CC-AP orders the HG 3575 to do a restart and to start up with the host CC.

For this reason, switchback should occur under controlled circumstances and at times of low traffic.

2.1.3.3 Remarks

Only the AP of the 3700-IP type can be equipped with the CC-AP

The AP Emergency Option needs, in addition to the CC-AP hardware, hardware installed into the AP 3700 IP:

– cPCI cassette with backplane and fan
– DSCXL-Processor Card
– Harddisk module
– Power Supply module

Restrictions

The following restrictions of the feature "AP Emergency" have to be considered:

- The cloned database at a CC-AP cannot be in sync with the host database at any time. The implementation allows a manually triggered and a cyclic update e.g. once a day. Every change in the database of the host system applied between the updates is not available at the CC-AP and thus lost in case of AP emergency. This includes both administered changes and changes applied by the user, like timer, speed dial targets, call forwarding targets, etc.
- Applications which use CAP to monitor/control calls in the HiPath 4000 system are supported in emergency operation also. The CAP can monitor resources in the host system and on the CC-APs. This requires that the IP connectivity between CAP and CC-AP is available in the emergency situation.
- Applications that require availability of specific resources (IVR units, announcements …) can only work, if these resources are controlled by the same CC-AP that handles the rest of the call.

For the AP emergency feature the signaling connections are monitored. A switch over to AP emergency mode depends on the loss of these connections. A poor payload quality is processed by the payload survivability feature.

In AP emergency mode not all normal operation features can be supported.

2.1.4 Disaster Recovery

The feature „Disaster Recovery“ offers enhanced survivability functionality. It is possible to install a second HiPath 4000 system that in the case of a disaster (e.g. fire, water damage,...) is able to take over the functions of the main system.

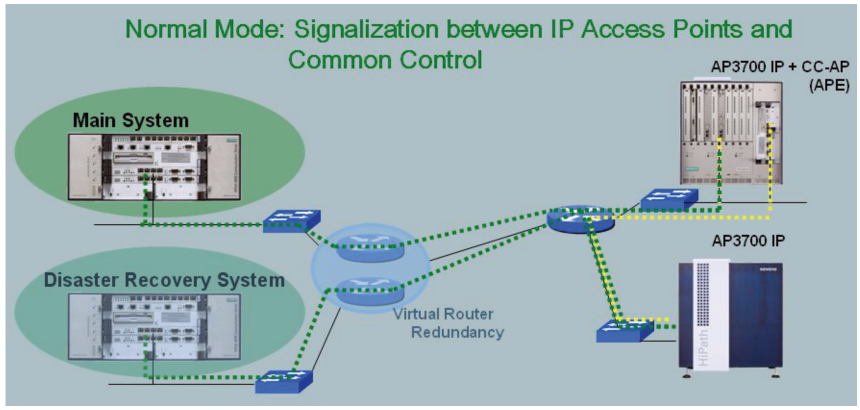


Figure 10 *Normal mode signaling and synchronization of main system and disaster recovery system*

If the main system is destroyed, the connected IP access points detect that the signaling to the main common control (CC-A / CC-B) is not possible and will fallback automatically to the configured AP-Emergency Mode.

The disaster recovery system will be startet during the AP emergency mode is still active. If the common control of the disaster recovery system is active and reachable for all AP3x00 IP in the customer network then the AP3x00 IP will switch back from CC-AP to the disaster recovery system.

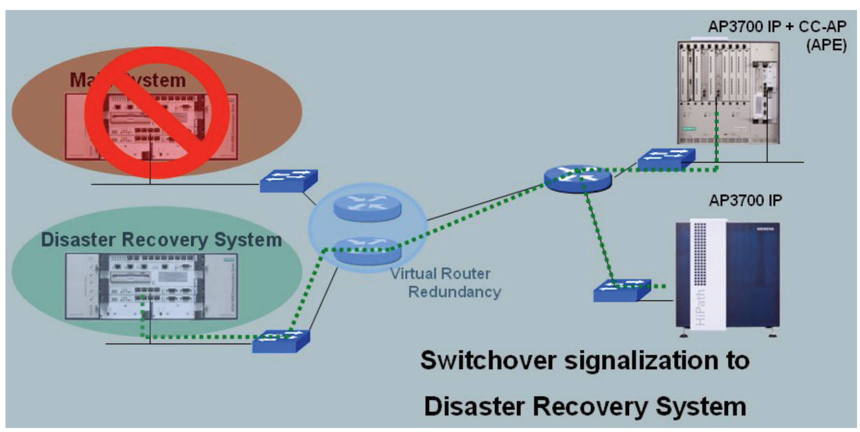


Figure 11 *Signaling after switch over to disaster recovery system*

In combination with AP-Emergency and the HiPath 4000 Duplex capability, this Disaster Recovery functionality guarantees the high availability of the HiPath 4000 system. The following table gives an overview of possible failure situations and the appropriate HiPath 4000 Survivability solutions.

Failure	HiPath 4000 Survivability Solution	Standby-Mode
Hardware fault in processor board	Duplex Call Control	Hot Standby
Soft/Hard Restarts	Duplex Call Control	Hot Standby

Failure	HiPath 4000 Survivability Solution	Standby-Mode
WAN failure for remote branches	Signaling Survivability	Hot Standby
	AP-Emergency	Warm Standby
Complete failure of main system	AP-Emergency (automatic activation but survivability for maximum 30 days)	Warm Standby
	Disaster Recovery (activation by administrator and und no time limit on survivability)	Cold Standby with synchroniz. database/ soft-ware

2.1.4.1 Prerequisites

- Two HiPath 4000 systems (main system and disaster recovery system).
- Access Point Emergency in order to connect the access points in the customer LAN and to the HiPath 4000 main system (TCP/IP connection).
- Only IP access points (AP3x00 IP) are supported.
- Subscribers directly connected to the main system can no longer be used in case of disaster recovery mode.
- Applications redundancy is required (e.g. ProCenter using Atlantic LAN)
- The main and disaster recovery system has the same hardware and software version
- There must be an interconnection (C-LAN) between main system and disaster recovery system.
- The common controls of main system and disaster recovery system must have the same IP address, so that both common controls will address the same broadcast domain.

2.1.4.2 Remarks

- Disaster Recovery does not replace any type of data backup.
- Subscribers directly connected to LTUs Shelves of the main system can no longer be used in case of disaster recovery mode. These subscribers need to be moved to other shelves, which are connected directly to the disaster recovery system.
- The Unix part is synchronized by data backup. From main system a backup is periodically transferred to an FTP server. The disaster recovery system periodically checks if new backup is available on FTP server.

– Disaster Recovery is only released for cPCI hardware.
– The disaster recovery system will not be registered at the HiPath 4000 Manager.
– The switch over must be done manually

2.2 IP Distributed Architecture

2.2.1 Description

HiPath 4000 provides the possibility of distributing Access Points via an IP network. The user connection to the access points will be treated as if they are connected directly to the HiPath 4000 system in the traditional way.

The main characteristics of the IPDA concept are:

– the control is done by exchanging all signaling messages (D-channel information) between the HiPath 4000 and the access points via IP network.
– the voice/data media stream (B-channel information) is exchanged directly between the access points via IP network.
– this allows to connect the access points independently of their location.

There are three main components in the IP distributed architecture scenarios:

– the communication server that takes control of all components of the HiPath 4000 distributed architectue. The entire amount of telephony features is available in the IP distributed architecture.
– at least one HiPath HG 3500 module is needed to establish a payload connection between subscribers/trunks of the HiPath 4000 and subscribers/trunks attached to the access points.
– IP Access Points
  There are two types of IP based access points:
  – The HiPath AP 3300 IP offering 16 slots
  – The HiPath AP 3700 IP, offering up to 9 slots in 19" architecture.

  Both types of access points are equipped with HiPath HG 3575 as central control device providing the connectivity to the IP infrastructure (10/100 BaseT).

HiPath 4000 integrates mechanisms needed to allow high voice quality for IP-based access points. These mechanisms include:

– Echo compensation

- Option for bandwidth reduction
(Voice Compression G.729A with 8 kbps, and Silence Suppression)
- Quality of Service support via IP network by traffic prioritization:
IEEE 802.1 d/q and IETF DiffServ

2.2.2 Distributed Switching

With HiPath 4000 the switching of payload channels (B-channels) is no longer limited to the central switching matrix, because:

- Calls within IP based access points are switched with no delay in the local TDM-based switching matrix.
- Calls between IP based access points are routed within the IP Network
- Calls not limited to IP based access points are routed both in the IP network and HiPath 4000

Signalling and payload survivability ensure that in a distributed environment HiPath 4000 still provides highest availability. The PSTN can be used as a backup network if the IP network fails or does not provide the quality required for voice.

Restrictions

Not supported standards are H.323, H.450, ECTF(Enterprise Computer Telephony Forum)

2.2.3 Basic conditions for VoIP

Using IP networks for real-time traffic such as voice imposes some Quality-of-Service (QoS) requirements that are not essential for data traffic. Real-time traffic has specific requirements concerning:

- Delay
- Delay variation (jitter)
- Packet loss
- Bandwidth availability and efficiency

Further requirements don't differ much from the requirements already established in data networks (e.g. reliability, security) and are usually already well enforced in todays data networks (e.g. firewalls).

Delay

The design goals of ITU (International Telecommunication Union) for end-to-end delay are:

- < 150ms One-way delay (target)
- < 400ms One-way delay (acceptable)

Jitter

The jitter buffer is used to compensate delay variation (jitter). The jitter buffer size should be choosen so that it can compensate two times the delay introduced by the codec.

Packet loss

The design goals for packet loss and delay are dependent on each other: The more packet loss in the network the lower the acceptable delay and vice versa. To achieve acceptable voice quality the maximum packet loss should be lower than 1% to 3% (dependent on delay).

Bandwidth

Bandwidth availability and efficiency is essential especially as bandwidth often is a limited resource and has cost connotations. Voice compression and Voice Activity Detection (VAD) with Silence Suppression are known measures to reduce bandwidth requirements.

But also bandwidth usage and delay are dependent on each other. The delay increases when an IP packet with a certain fill level is filled with bandwidth-reduced, e.g. compressed, voice. In addition bandwidth usage decreases, for example, when the IP packet fill level increases (and vice versa).

The bandwidth reduction possible when using VAD varies dependent on factors like intensity of background noise. Thus it is not possible to state an exact value for bandwidth reduction.

Potential action to take in the IP network

Before introducing voice onto the IP network it has to be made capable of supporting the QoS required for voice. Siemens offers the appropriate professional services to support this important consideration.

- Adequate dimensioning and structure of the network (e.g. for low packet loss rate, low delay
- Deployed LAN components should be Ethernet switches; shared hubs may cause problems.
- Implementation of guaranteed service levels, typically defined in a service level agreement with a carrier or service provider. These guaranteed service levels can be achieved using specific mechanisms, such as: traffic prioritization on different levels
(e.g., IEEE 802.1d/q, IETF DiffServ).

2.2.4 Interconnect capacity for IP Solutions

The Gateways for IP Distributed Architecture (IPDA), HG3500 and HG 3575 provide an extended B-channel capacity. Scalability is provided with a low-end version supporting up to 60 B-channels and a high-end version supporting up to 120 B-channels (depending on encryption, components, ...).

LAN interface

The HG 3575 board is equipped with two 10/100 Base-T interfaces. The second LAN interface can be used to doubly secure the AP to the IP network. Switchover only takes place in the case of layer 1 errors.

V.24 Interface

On the HG 3575 board two V.24 interfaces classified as DTE are provided. One interface is used for CLI service interface. The other interface is used for remote survivability via modem. The baud rates are programmable up to 115200 bps

SIU (HG3575 only)

The DSP based part of the signaling unit provides the following functions which are necessary in Access Points:

– Tone generation
– DTMF receiver/transmitter
– Music-on-hold
– Dialtone receiver
– Announcements
– Trunk diagnostic system function
– Test functions

2.2.5 Voice Codecs

The following voice codecs and features are provided with IPDA in HiPath 4000.

Module	Feature	Comment
VOICE		
Codecs	G.711A-law (TDM)	Hicom internal coding is each case A-law
	G.711A-law (LAN)	
	G.711µ-law (LAN)	
	G.729AB	VAD, CNG, PLC

Table 1 Voice codecs

Module	Feature	Comment
	G.711 Annex I	PLC
	G.711 Annex II	VAD, CNG
RTP (LAN-Side)	RTP Send	IETF RFC 1889
	RTP Receive	IETF RFC 1889
	RTCP	RTCP protocol in Host
	Configurable Jitterbuffer	
	DTMF over RTP	Required for reliable DTMF –transport, because there is no H.323/H.245 available
Telco (ISDN-Side)	DTMF Send	ITU Q.23
	DTMF Receive	ITU Q.24
	LEC G.165/G.168-48ms echo tail length	
	Fax-Tone Detection	CED (2100Hz), CNG (1100Hz) tone
	Modem-Tone Detection	ANS, ANSbar tones
FAX	Transparent Fax	Faxes are transported over a G.711 channel (LEC, DTMF switched off) For low speed fax <= 14400 bit/s, the LEC is still activated with all non-linear components disabled. For high speed fax the LEC is disabled and 6dB attenuation will be applied. PLC and VAD will be disabled in both cases.
MODEM	Transparent Modem	Modem calls are transported over a G.711 channel (LEC, DTMF switched off) For half duplex modems, the LEC is still activated with all non-linear components disabled. For full duplex modems the LEC is disabled and 6dB attenuation will be applied. PLC and VAD will be disabled in both cases.
CLEAR CHANNEL		bit transparent mode for special ISDN dataservices, e.g. H.320 videoconferencing over IP-trunks

Table 1 Voice codecs

The HG 3500/3575 boards provide the following voice codecs (on the LAN-side):

G.711

The two G.711 variants (G.711 A-law and G.711 µ-law) are supported. The minimum frame size is 10ms.

G.729AB

The G.729 algorithm is used for speech coding at 8 kbit/s using the CS-ACELP mechanism. The Annex A indicates that the reduced complexity 8kbit/s codec is used and Annex B describes the VAD/DTX and CNG algorithm.

- Operates on speech frames of 10ms corresponding to 80 samples at the sampling frequency of 8000 samples/sec.
- Voice Activity Detection VAD: ON or OFF

Currently, the minimum encoding time that can be configured by AMO is 30 ms for G.711 and 20 ms for G.729AB.

2.2.6 Signaling and payload separation

Signaling and payload separation is a feature which enables HiPath 4000 in an IPDA- environment to split the path of the network traffic in the meaning of

- signaling from Hostsystem (call control) to Access Point is routed over the IP Customer Network (Enterprise WAN)
- payload is routed over the PSTN Network using payload survivability connections.

Signaling and payload separation based on existing Survivability solution and therefore the signaling path is routed in the same way as today. The payload (RTP-Stream) is routed across connections built up from WAML or ISDN-Router via PSTN-Lines between the locations.

Signaling and payload separation can be configured in a way that

- payload separation is always active (static signaling and payload separation)
- payload separation is used in case of limited WAN- bandwidth managed by the resource manager or overflow scenarios when then maximum of bandwidth is reached (payload overflow)
- payload separation is used in case of network failure (emergency routing)

2.2.6.1 Payload separation is always active

In this case a static signaling and payload separation is configured. All signaling information are routed via the enterprise WAN to each location respectively acces point. All payload connections are routed via ISDN- connections to the PSTN.

Several possibilities are available when payload separation is always active:

– Channel limitation in access points
– Selective access if a privileged subscriber specifies via access code that he does not wish to use IP-network

This solution requires:

– Payload survivabilty to all access points.
– All access points have to be assigned in different source groups.
– Transfer trunk to trunk is mandantory.

2.2.6.2 Payload separation in case of overflow

Signaling and payload separation is used in case of limited bandwidth over WAN connection managed by Resource manager or overflow scenarios when maximum of bandwidth is reached. In case that bandwidth limit is reached overflow will be made over ISDN lines.

The following cases are possible when control is performed with Resource Manager:

– IP- phones in a Access point behind WAN connection with limited bandwidth
– Access Points which are connected via WAN with limited bandwidth

Resource Manager limits number of calls. In case that more calls are made that are permitted, not all calls can be answered.

This solution requires:

– Payload survivability to all access points.
– All access points have to be assigned in different source groups.
– All calls between different access points will be made over ISDN-lines.
– Transfer trunk to trunk is mandantory.

2.2.6.3 Payload separation in case of emergency

Signaling and Payload Separation may be also used in emergency situation. For example if the IP connection between tthe central HiPath 4000 switch and an access point fails.

This solution requires:

– Payload survivabilty to all access points.
– All access points have to be assigned in different source groups.

- Transfer trunk to trunk is mandantory.
- All calls between different access points will be made over ISDN-lines.
- Signaling survivability has to be established and activated.

2.2.6.4 Supported Features

Subscriber features

- Basic call
- Hunt group
  (all members has to be located in one source group respectively one Access Point))
  - Forwarding for Hunt group members in different source groups is not supported
- Pick up Groups
  (all members has to be located in one source group respectively one Access Point)
  - Group call is not supported
  - Pick up is possible from analog telephone only if the called party is on the same AP
- Call back
- Call forwarding
- Transfer and toggle
- Conference
- Keyset functions (without Phantom lines)
  (all members has to be located in one source group respectively one Access Point)
  - Keysets cannot pickup call from a secondary line a call over SPS
  - Keyssets from different source group cannot be retrieve call from hold
- Name Display
- Number Display
- One number Service - ONS-
  (all members has to be located in one source group respectively one Access Point)
- Call Waiting

– Call Log
– Message Waiting Integration
– OpenScape Xpressions Connectivity
– Direct station selection - DSS - (planned)
– Camp on (planned)
– Override (planned)

Remark: For these features the user experience and functionality will differ to the user experience, when the call is forwarded over LAN/WAN connections.

2.2.6.5 General Prerequisites:

– Central office connectivity via ISDN- lines
– Signaling and payload connections has to be built up via WAML or external ISDN-Router
– Bandwidth limitation has to be performed by the Resource Manager.
– SPS is restricted to star topology/IPDA-infrastructures.
– Using HG 3500 separate boards are needed for trunking and WAML functionality.
– Currently only 8 static routes can be configured on the HG 3500. Therefore the number of subnets for the APs is limited. In the worst case (each AP in another subnet, no combination of subnets via the netmask possible) this limits the amount of AP's using SPS without an external router to 8. In addition only 8 WAMLs can be used in the host.
– Delays in signaling may cause wrong interpretation with a result of wrong call treatment and can affect the features behavior (e.g. Hunt group, call pick up groups)
– Due to the fact, that in call pickup scenarios there is no connection before the picking user performs the pickup, cut trough times might be longer in SPS scenarios
– Payload separation is destination oriented and restricted to one HiPath 4000 system. If other systems are connected to the HiPath 4000 no networking features will be supported (e.g. network wide pick-up).
– Different trunk groups must be configured for payload separation without intercept parameters.

2.2.6.6 Restrictions

- VNR is not supported
- Path replacement is not supported.
- Mobile HFA is not supported.
- In the case of overflow, attendant console functions are not support.
- Multiple queuing at Attendant console is not supported.
- Synchronized announcements are not supported.
- MLPP not supported.
- AP Emergency is not supported.
- Not all devices supports the listed feature set (e.g. SIP-telephones ,ISDN-devices , DECT-telephones. The feature set is restricted only to basic call functionality.
- ACD/ACL is realized in the "Payload survivability " concept because the complete ACD/ACL features support cannot be guaranteed.
- Group call constellations like Chese Keyset, Pick up Groups, Broadcast, ACD, Park, override functions,… are not allowed when the members are in different source groups.

2.3 Common Gateway Architecture

2.3.1 Multiple Feature Support

HiPath 4000 “Common Gateway Architecture” supports various functionalities (multiple feature support) based on one hardware board. Within one hardware, the software of the HiPath Gateway 3500 supports:

- IP- subscriber (HFA)
- SIP- subscriber
- SIP- Q- trunking
- Nativ SIP- trunking
- IP- distributed architecture (host-side payload connectivity to access points)
- WAML- functionality

A mix of the mentioned functionality is also running on the same board in parallel, but nevertheless the usage of several HG 3500 boards is possible too and in some scenarios necessary.

Additional functions

– In general the coder G.711, G.723.1, G.729 and their options are provided. Only IP- subscriber and IP- Trunking mode supports the codec G.723.1. Only one type of codec (it is a codec set with priorisation) per board could be configured.

– Signalling and Paylod encryption based onTLS V1.0 (Transport Layer Security according to RFC 2246, 2712, 2817, 2818, 3268) and SRTP (Secure Real-Time Transport Protocol according to RFC 3711) with Advanced Encryption Standard (AES).

– For Signaling and Payload Encryption the PKIDC supports the generation of the certificates by DLS-PKG or the import of the certificates from a customer PKI. Beside the delivery of the certificates PKIDC supports also the distribution of Security Profile parameters as well as the CRL distribution point in case an external PKI is used.

The following details have to be noted:

– On one HG 3500 board a maximum of 120 B-channels are available.

– Granularity of B-channels for IP- subscriber mode is “1”.

– Granularity of B-channels for SIP- trunking mode, IP- distributed architecture (host side) is “10”.

– Granularity of B-channels für WAML- functionality is “30”.

– For each circuit assigned to either SIP-Trunking mode or WAML functionality logically 30 B-channels are reserved.

– A maximum of four circuits can be configured for SIP-Q- trunking mode at one board.

– Overbooking is only implemented for IP- subscriber mode.

– VoiP Security with Signalling and Payload Encryption is supported

– Codec G723.1 isn’t supported in IP- distributed architecture mode

Compatibility to gateways in former versions

HG 3530, HG 3550 and HG 3570 boards are not supported within HiPath 4000 V5.

HG 3500 is compatible to the former HG 3530/ 50/70 V2.0 and V3.0 ones only for basic functionality.

Common gateway interoperation to HiPath 4000 V3.0 gateway boards will only work, if security mechanisms are disabled. E.g. security features will only work within pure installations.

2.3.2 IP- Subscriber Mode (HFA) and mobile HFA

Within the IP- subscriber Mode, IP-Phones support the HiPath 4000 feature set in the so called HFA-Mode (HiPath Feature Access).

- It is possible to configure a mix of SIP and HFA subscribers with a granularity of 1 channel.
- HiPath IP-Subscriber mode (HFA) supports in addition the feature "mobile HFA".
- Overbooking up to 240 subscribers are possible
- In addition the IP-subscriber Mode supports the mobile HFA- function

Mobile HFA

The "mobile HFA-logon-feature" allows a mobile telephone user to use all functions and features at a visited HiPath location inclusively the user interface as at the home station. Prerequisites of "mobile HFA-logon" are:

- Both phones (home phone and visited phone) have to be a HFA IP phone and must support "mobile HFA logon"
- Both phones (home phone and visited phone) must have a display
- It is recommended that both telephones have the same equipment / key layout.
- If differencies occur, such functions will be transferred to the visited phone which can be shown at the device.

Parameters which depend on the location of the phone or define capabilities of the phone are unchanged when using the phone by a mobile HFA IP user. E.g: DHCP on/off, IP address, subnet mask, default gateway, G.711/G.723/G.729. These data have to be administered independent of HFA logon.

A station which can be used by a mobile HFA-user has to be activated with a logon to a base home access if the phone isn't used by a mobile HFA user. This is necessary for supporting the user interface for activation "mobile HFA user" and support of service and emergency calls.

The IP address and HFA password of the home base access have to be stored as default in a visited HFA IP phone as in a deactivated home IP phone. This is necessary to enable again Logon to the respective home accesses after deactivation of a mobile HFA IP user by "mobile HFA Logoff". In addition to this "home address" the temporary home address, phone number and password of the mobile HFA user have to be stored in a HFA IP phone.

If an analogue adapter is installed at the home and the visited phone the Logon at the visited phone implies the logon of both (phone and adapter) at the home station. If there's no analogue adapter at the home or visited station, only the phone logs on.

Scenario A: Basic Access of an IP Phone

The activation of logon/logoff by the HFA pone with the already implemented functionality is supported unchanged.

The data which are handed for the connection to the switching system (IP address of the home board, HFA password etc.) are administrated by a local IP phone procedure.

After logoff the home base address (as well as the possibly temporary stored home address) of a mobile HFA user are erased from the HFA IP phone.

For an office phone the assignment to the switch is directed to a home address which is administrated with the data and functions of the station owner. For a shared desk phone the assignment to the switch is directed to a base address which is administrated with a help number with respectively low class of service.

If a HFA IP phone activates a new Logon (basic Logon) with valid phone number and valid password during an other HFA IP phone has been activated via a Logon with the same phone number and password, this phone is disconnected in each phone state (idle, call state, dialing, ringing, etc.) and is deactivated via Logoff. With the Logon of the new IP phone this phone is activated.

Scenario B: "Mobile HFA logon" at a visited Station

A mobile HFA user activates "Mobile HFA logon" at a visited station (e.g. at a shared desk) which is equipped with a HFA IP phone

Procedure and implementation steps (Access code - subscriber number - PIN). Alternatively to the access code the activation can be done via menue point of the service menue.

If a mobile HFA user has activated "Mobile HFALogon" and if therefor the visited HG3530 board is in the HFA Logoff state, all incomming calls to the base number of the shared desk phone are forwarded to that destination which is administered as Call Forwarding No Reply CFNR at the base number of the shared desk phone.

Special case: the home access of a mobile HFA user needs not to be equipped with an HFA IP phone - the mobile user without home phone. But in this case the home board HG3530 has to be equipped and the mobile user has to be administered.

Scenario C: Activation of "Mobile HFA Logon" at a second shared desk phone

A mobile HFA IP user activates "Mobile HFA logon" at a shared desk phone 2 after he/she was activated at a shared desk phone 1 before.

If a mobile HFA user has activated "Mobile HFA-Logon" and if therefore the visited HG3530 board is in the HFA Logoff state, all incoming calls to the base number of the shared desk phone 2 are forwarded to that destination which is administered as Call Forwarding No Reply CFNR at the base number of the shared desk phone 2.

Scenario D: Activation of "Mobile HFA Logon" at a second office phone

A mobile HFA IP user activates "Mobile HFA-Logon" at an Office Phone 2, after he/she was activated at his/her Office Phone 1 or an other visited HFA phone 1 before.

If a mobile HFA user has activated "Mobile HFA-Logon" and if therefore the visited HG3500 board is in the HFA Logoff state, all incoming calls to the home number of the Office 2 user are forwarded to that destination which is administered as Call Forwarding No Reply CFNR at the home number of the Office 2 user.

Scenario E: Re-Activation of an Office Phone

A mobile HFA user has activated "Mobile HFA-logon" at a visited HFA IP pone and returns to his/her deactivated Office pone.

If a mobile HFA user has activated "Mobile HFA Logon" at a visited HFA IP phone and didn't deactivate at this visited phone he/she is able to activate Logon (basic Logon) by a manual procedure to his/her own home access at his/her HFA office phone which is in the Logoff state. With that the connection of the up to now visited phone to the home access will be deactivated. The activation of this Logon procedure is a local function of the phone (simplified procedure for input of own number and HFA password).

For optional protection of this Logon activation which deactivates the connection of the before visited IP phone to the home access in each call state, a digit string has to be stored within the IP phone by administration. The Logon will be executed after dialing the digit string.

Scenario F: Deactivation of "Mobile HFA Logon" at the visited phone

After a mobile HFA user has activated "Mobile HFA logon" at a visited HFA IP phone he/she can deactivate at this visited station by "Mobile HFA logoff".

The mobile HFA user activates the "mobile HFA logoff" by procedure (Access Code or menue point of the service menue). The HFA office phone is reactivated automatically

Scenario G: Deactivation of "Mobile HFA Logon" via AMO

HiPath 4000 can deactivate the mobile user at a visited station via service command (AMO).

The deactivation via AMO is possible as well at the home node of the mobile HFA IP user as at the node of the visited HFA phone.

The home board HG 3500 of the mobile HFA IP user sends "Mobile HFA logoff" to the up to now activate

Restrictions

Within this feature "mobile HFA logon" no requirements are considered for

– routing of outcalls into the respective public network

– emergency calls into the respective public network

– changing time zones.

2.3.3 SIP- Subscriber

SIP devices are handled in HiPath 4000 as functional devices (like DSS1-phones). The main difference between functional devices and other devices (e.g. digite) is that while a digite is under absolute control of the switch (stimulus mode), a functional device must be considered a separate, independent call processing entity. The switch communicates with a SIP phone using SIP messages via HG 3500, but the switch cannot affect how the SIP phone will handle those messages.

Due to this fact there are differences between SIP- telephones and digital system- or IP- telephones in usage and functionality of the supported features.

2.3.3.1 Supported subscriber features

Depending on the used SIP-Phones the SIP- subscriber mode supports the following functions:

– Block dialing or time out dialing

– Calling Line Identification Presentation/Calling Line Identification Restriction

– Connected Line Identification Presentation/Connected Line Identification Restriction

– Call Hold

– Call Retrieve

- Call Toggle
- Call Waiting
- Call Transfer (attendant Transfer)
- Local Do not Disturb
- Local 3rd Party Conference
- Call forwarding
  - unconditional
  - no reply
  - busy
- Call deflection
- Name presentation
- Second Call
- Parallel ringing (ONS)

2.3.3.2 Supported system features

HiPath 4000 supports in general the following system features for SIP-subscibers:

- mix of SIP and HFA subscribers. Granularity of B-channels for IP-subscribers is “1” (not possible in conjunction with HiPath 4000 SoftGate)
- T38 Fax-Support (OSCAR profil)
- Member of an hunt group
- End-to-end payload connection
- Registration
- CSTA-Monitoring

2.3.3.3 Supported Request for Comments

- RFC 2198 - RTP Payload for Redundant Audio Data
- RFC 2327 - SDP: Session Description Protocol
- RFC 2396 - Uniform Resource Identifiers (URI): Generic Syntax
- RFC 2617 - HTTP Authentication: Basic and Digest Access Authentication

- RFC 2833 - RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC 3261 - SIP: Session Initiation Protocol
- RFC 3262 - provisional Response Acknowledgement (PRACK) Early Media
- RFC 3264 - An Offer/Answer Model with SDP
- RFC 3265 - Session Initiation Protocol (SIP)-Specific Event Notification
- RFC 3310 - HTTP Digest Authentication
- RFC 3311 - Session Initiation Protocol (SIP) UPDATE Method
- RFC 3323 - A Privacy Mechanism for the Session Initiation Protocol (SIP)
- RFC 3324 - Short Term Requirements for Network Asserted Identity
- RFC 3325 - Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks
- RFC 3515 - The Session Initiation Protocol (SIP) Refer Method
- RFC 3550 - RTP: Transport Protocol for Real-Time Applications
- RFC 3551 - RTP Profile for Audio and Video Conferences with Minimal Control
- RFC 3725 - Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP)
- RFC 3891 - The Session Initiation Protocol (SIP) Replaces Header
- RFC 3892 - The Session Initiation Protocol (SIP) Referred-By Mechanism
- RFC 4028 - Session Timers in the Session Initiation Protocol (SIP)

2.3.3.4 Remarks and restrictions

Please refer to the actual sales releases concerning the supported features.

- The usage of codec G.729 is restricted to the codec with the highest priority, although the WBM interface allows to choose all codecs (G.729, G.729A, G.729B and G.729AB)
- The following functions are not supported for SIP subscribers:
  - Exec. / Secr. - SIP terminals cannot be configured as Chese members.
  - Pickup group - SIP terminals cannot be members of a pickup group.
  - Keysets - SIP terminals cannot be configured as keysets.

- PIN and PIN-dependent features (e.g. Class of Service Changeover, Multiple FWD (Follow-me))
- Emergency override / emergency release
- System integrated call log
- Night server (only possible passively, not actively)
- Silent Monitoring - cannot be configured for an SIP terminal.
- Witness Facility - An SIP terminal can only be a passive subscriber for the witness facility.
- direct calling (DSS).
- Park in system - An SIP subscriber cannot actively park another subscriber in the system. However, the SIP subscriber can be parked by another subscriber.
- Malicious Call Identification/Tracing - An SIP terminal can be traced, but cannot trace a connection itself.
- Override.
- Callback free / busy - A callback can be entered on an SIP terminal if free / busy, but the SIP terminal itself cannot enter a callback.
- Remote Call Forwarding.
- Speed dialing and individual speed dialing - Cannot be programmed or performed by an SIP terminal.
- Display suppression.

2.3.4 SIP- Q Trunking, networking to HiPath systems / OpenScape Voice

CorNet IP will not be longer the preferred IP-trunking protocol in HiPath 4000 networks. HiPath 4000 V5 requires SIP-Q as the only trunking protocol:

- in homogeneous HiPath 4000 networks (V4 and higher)
- connectivity to HiPath 2000 V2 and higher
- connectivity to HiPath 3000 V7 and higher
- connectivity to OpenScape Voice V3.1 and higher

The „SIP-Q Trunking“ feature uses procedures described in the ECMA 355 standard to connect HiPath systems via IP based networks. CorNet NQ messages and CorNet NQ features are tunneled over SIP-Q.

The available features interdepends directly from the connected communication system. Therefore please refer to the actual sales releases concerning the supported features.

Remarks and restrictions

– The usage of codec G.729 is restricted to the codec with the highest priority, although the WBM interface allows to choose all codecs (G.729, G.729A, G.729B and G.729AB)

– Only one trunking protocol can be set up on an interface card (SIP-Q-Trunking, CorNet IP or Native SIP-Trunking).

2.3.4.1 Networking in homogeneous HiPath 4000 networks

In homogeneous HiPath 4000 networks SIP-Q-Trunking, supports all further network wide features and function known from the protocols CorNet NQ respectivly CorNet IP.

Remarks and restrictions

– CorNet IP in conjunction with HG 3500 will be still supported für backwards compatibility with HiPath 4000 V2.0/V3.0

2.3.4.2 Networking to other HiPath platforms or OpenScape Voice

Basically HiPath 4000 supports the following networking features with OpenScape Voice:

– CLIP / CLIR
– COLP / COLR
– CNIP/CNIR
– CNOP/CNOR
– DTMF
– Name display
– Hold / Retrieve / Toggle
– Call transfer
– Call forwarding (CFU/CFB/CFNR)
– Call deflection
– Second call

– Callback on busy subscriber / Callback No Reply
– Do Not Disturb
– Digest Authentication
– End-To-End Payload
– Message Waiting Indication
– Route Optimization
– HiPath 4000 as Survivability Media Gateway
– Signaling and Payload Encryption (TLS / SRTP)
– T38 fax (OSCAR profil)

2.3.4.3 Supported Request for Comments with SIP-Q-Trunking

Basically HiPath 4000 supports ther following Requests for Comments

– RFC 2198 - RTP Payload for Redundant Audio Data
– RFC 2246 - The TLS Protocol Version 1.0
– RFC 2327 - SDP: Session Description Protocol
– RFC 2396 - Uniform Resource Identifiers (URI): Generic Syntax
– RFC 2617 - HTTP Authentication: Basic and Digest Access Authentication
– RFC 2833 - RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
– RFC 2976 - The SIP INFO Method
– RFC 3204 - MIME media types for ISUP and QSIG Objects
– RFC 3261 - SIP: Session Initiation Protocol
– RFC 3262 - Provisional Response Acknowledgement (PRACK) Early Media
– RFC 3264 - An Offer/Answer Model with SDP
– RFC 3310 - HTTP Digest Authentication
– RFC 3311 - Session Initiation Protocol (SIP) UPDATE Method
– RFC 3326 - The Reason Header Field for the Session Initiation Protocol (SIP)
– RFC 3550 - RTP: Transport Protocol for Real-Time Applications

- RFC 3551 - RTP Profile for Audio and Video Conferences with Minimal Control
- RFC 3711 - The Secure Real-time Transport Protocol (SRTP)
- RFC 3725 - Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP)
- RFC 3830 - MIKEY: Multimedia Internet KEYing
- RFC 4028 - Session Timers in the Session Initiation Protocol (SIP)

2.3.5 Native SIP- Trunking

Nativ SIP- Trunking supports the following functions:

- native SIP: support for SIP- carrierconnection
- other SIP-based systems (e.g. 3rd party application like conference server, Voice Mail Server)

For further informations please refer to the actual sales releases concerning the supported features, providers and 3rd party applications.

Remarks and restrictions

- The usage of codec G.729 is restricted to the codec with the highest priority, although the WBM interface allows to choose all codecs (G.729, G.729A, G.729B and G.729AB).
- Only one trunking protocol can be set up on an interface card.
- HG 3500 must be configured as a remote gateway to SIP service provider (no LEGK function).
- No DNS support.
- NAT traversal, STUN and PPPoE to service provider are not supported by the SIP trunk gateway.
- No DMC support.
- No MSN support.
- No simultaneous SIP provider support on one gateway but the support of different SIP provider profiles per board is possible.

2.3.5.1 Supported features

Basically HiPath 4000 supports the following features:

- Block dialing or time out dialing

- Calling Line Identification Presentation/Calling Line Identification Restriction
- Connected Line Identification Presentation/Connected Line Identification Restriction
- Call Hold
- Call Retrieve
- Call Toggle
- Call Waiting
- Call Transfer
- Conference
- Call forwarding
- Name presentation
- T38 Fax-Support (OSCAR profil)
- Message Waiting Indication
- DNS Client support
- DNS Failover support (Timer T1/T2) / DNS_SRV
- Authentication supported for
  - Gateway device-authentication (REGISTER method) and
  - SIP session-authentication (INVITE method)

2.3.5.2 Supported Request for Comments with native SIP-trunking

Basically HiPath 4000 supports ther following Requests for Comments

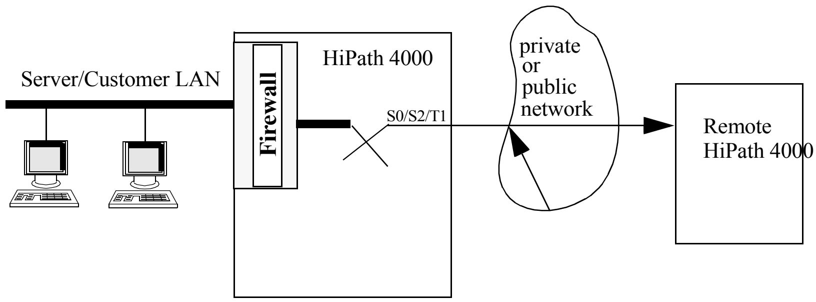
- RFC 2198 - RTP Payload for Redundant Audio Data
- RFC 2327 - SDP: Session Description Protocol
- RFC 2396 - Uniform Resource Identifiers (URI): Generic Syntax
- RFC 2617 - HTTP Authentication: Basic and Digest Access Authentication
- RFC 2833 - RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC 3261 - SIP: Session Initiation Protocol

– RFC 3262 - Provisional Response Acknowledgement (PRACK) Early Media
– RFC 3263 - SIP Locating Servers
– RFC 3264 - An Offer/Answer Model with SDP
– RFC 3310- HTTP Digest Authentication
– RFC 3311 - Session Initiation Protocol (SIP) UPDATE Method
– RFC 3323 - A Privacy Mechanism for the Session Initiation Protocol (SIP)
– RFC 3324 - Short Term Requirements for Network Asserted Identity
– RFC 3325 -Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks
– RFC 3515 - SIP- Refer method
– RFC 3550 - RTP: Transport Protocol for Real-Time Applications
– RFC 3551 - RTP Profile for Audio and Video Conferences with Minimal Control
– RFC 3581 - An Extension to the Session Initiation Protocol (SIP) for Symmetric Response RoutingRFC 3324 - Short Term Requirements for Network Asserted Identity
– RFC 3891 - SIP-Heades "Replaces Header"
– RFC 3892 - SIP- "REFERRED-BY- mechanism
– RFC 4028 - Session Timers in the Session Initiation Protocol (SIP)

2.3.6 WAML functionality

Scenarios

1. The LAN interface for coupling a server/customer LAN with the private/public ISDN network.
   This interface provides an integrated LAN/ISDN router function in order to:
   a) reach the remote HiPath 4000 node from the server/customer LAN via ISDN (remote administration)
   b) reach the server/customer LAN remotely via ISDN (administration).

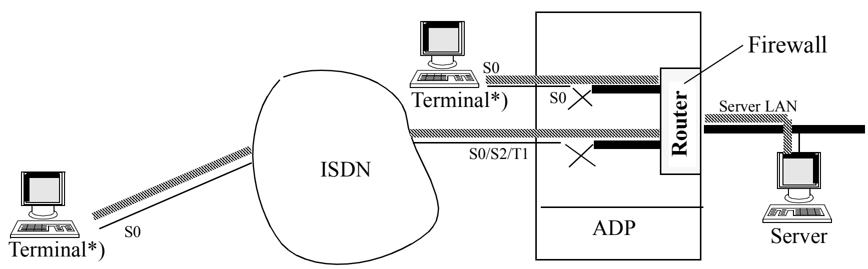


2. Remote access to the ADP

   A server/customer LAN connected via ISDN at the ADP of a remote HiPath 4000 node in the network.

   – A router function is required in the local system, in order to route the LAN connection to the ISDN output connection (S0 connection to the remote system).

   – A router function is also required in the remote system, in order to route the ISDN input connection to the ADP (Atlantic-LAN). A "firewall" prevents unauthorised access to the server LAN via ISDN.



*) Terminal emulation on PC

Scenario 2 illustrates the connection of remote terminals to the server (e.g HiPath 4000 Manager).

– A router function is required in the HiPath node, in order to route the ISDN input connection to the server at the server LAN. A "firewall" prevents unauthorised access to the server LAN via ISDN.

IMPORTANT: This routing functionality of the HG 3500 is only released for access to the HiPath 4000 Manager.

Protocols

- The TCP/IP protocol (incl. UDP) is supported for coupling LANs with HiPath 4000 :
  - on the LAN side TCP/IP
  - on the ISDN side: PPP protocol with TCP/IP in the B channel transparently (PAP/CHAP)
- CAPI compliant S0 cards (for PPP protocol) are supported for S0-LAN coupling.

Bandwidth on Demand

Several physical connections can be set up between two HiPath nodes and more than 64 kbit/sec. data transmitted. Dynamic switching of additional B channels (channel bundling) permits this upon reaching a defined threshold in the buffer for data packets.

"Channel bundling" can be configured

- per ISDN destination number,
- for the max. number of B channels,
- for the max. and min. threshold of the data buffer for adding and reducing B channels.

Firewall

A filter function is required to control the access via the HG3500 from

- Source check
  - The number of the caller (subscriber authentication) is checked in order to prevent unauthorised connections from external sources via ISDN.
  - IP and/or MAC addresses (can be configured) of hosts at the external server are chekked.
  - extended IP-firewall functions. In this case only 30 IP-trunking channels are support (resticted via AMO)
- Destination check of authorised sources as to whether
  - Access to ISDN is permitted
  - Access to LAN is permitted
- Callback
  An optional callback function prevents access via manipulated call numbers. This function releases (clears down) the incoming connection and automatically sets up a new connection with the configured call number.
  In order avoid endless loops with a HG3500 to HG3500 connection via the public exchange, a callback state is maintained in the HG3500 initiating the

connection request. In this state, the next incoming connection which is expected after connection release is the callback with the call number of the desired HG3500, and in this case a callback reaction is not carried out. The callback function should be set up to provide as many callback call numbers as there are defined call numbers.

Short Hold

"Short hold" refers to automatic connection release after a configured time period during which no data have been transmitted, respectively an automatic connection setup ("dial on demand") is carried out upon receipt of new messages for the given released connection.

The physical connection is set up again to the still existing logical connection, i.e. processing can continue without a new LOGIN. Due to ISDN’s fast connection setup time, a delay is hardly noticable (1 - 1.5 sec.). If the callback function is set, a callback response is carried out for each new connection setup in the called router (doubles delay).

Restrictions

- Time-out can be set per B channel (destination address).
- The duration of the metering pulse is not taken into account for determining the time after the unused connection has been released because metering pulses are no longer transmitted in ISDN. Call charge messages with accumulated costs or units are transmitted instead. The point in time for releasing the unused connection can not be precisely determined on the basis of this information.
- Broadcast messages do not reset the time-out, as connection release could never be carried out otherwise.

Call Charge Allocation

If the HG3500 sets up an ISDN trunk connection, the incurred call charges are allocated (assigned) to the call number of the B channel. Data can be transferred in both directions as long as this physical connection exists. If a connection exists from HG3500 A to HG3500 B via the public exchange, all devices connected to the LAN of HG3500 A and B can use the existing connection. The call charges are allocated to the call number of the B channel of HG3500 A which set up the connection. It is not possible to allocate the call charges to the individual devices or applications connected to the LAN.

If the physical connection is cleared down by the short hold function, the connection is set up again as soon as new messages arrive. If these messages are data transferred from WAML B to A, i.e. HG3500 B has initiated connection setup, call charges are then allocated to HG3500 B. If the callback function is activated, the call charges are allocated to the called WAML because this HG3500 releases the incoming connection and actively sets it up again.

Not supported functions

- Data compression
- ISDN nailed connection
- Dynamic routing using router protocol
- SNMP interface, as administration and fault management is handled via the HiPath system interface
- Physically the 2nd LAN interface is available at the HG 3500 board. But this is already in use for WAML, which requires two active LAN interfaces (customer LAN and ATLANTIC LAN). This means, that a board with WAML functionality does not work together with the redundancy feature! If WAML functionality is configured there is no LAN redundancy at all.

2.3.7 Fax / Modem / DTMF

2.3.7.1 Fax-/Modem Übertragung and DTMF Erkennung

Within HiPath 4000 IPDA architecture Fax, Modem and DTMF tones are detected automatically on the new hardware.

Tone detection is done at the TDM side of the VoIP codec and is located before the nonlinear processing of the line echo canceller. Only the ANS/CED tone generated by the called party (Fax or Modem device) is evaluated. When such a tone is detected, the current codec (G.7xx) is changed to a special "G.711 F/M" codec which is optimized for analogue data and used for fax and modem transmissions.

For all half-duplex (hdx) devices (low-speed faxes 14,400 bps and hdx modems) the
G.711 F/M codec still needs echo cancellation but modified in a way that all nonlinear components are switched OFF.

For all full-duplex (fdx) devices (high-speed faxes and modems >14,400 bps) the line echo canceller is disabled and replaced by an attenuation of 6 dB.

In both cases, the jitter buffer is configured for data mode; furthermore, packet loss concealment and voice activity detection must be disabled.

An overview of the detected tones and the discrimination between hdx and fdx is shown in the next table

Tone	Explanation	Frequency	Operating mode
CED	T.30 Fax Answer	2100 Hz	hdx
ANS	Modem Answer	2100 Hz	hdx

Tone	Explanation	Frequency	Operating mode
ANS AM	Modem Answer	2100 Hz w/ V.8 amplitude modulation	hdx
ANS PR	Modem Answer	2100 Hz w/ V.25 phase reversals	fdx
ANS PR AM	Modem Answer	2100 Hz w/ V.25 phase reversals and V.8 amplitude modulation	fdx

Restrictions

- The fax detection module is only available in Voice Mode .
- T.38 Fax is only available in IP-trunking mode with CorNet IP NQ.
- A detection of V.21(1650 Hz and 1850 Hz) and demodulation of DIS sequence is not provided.
- The implemented solution for Fax/Modem Handling does not cover 100 % of all fax and modem devices; e.g. the following fax machines cannot be supported:
  - Some Group 3 fax machines which operate in manual mode, because they are allowed to skip the CED sequence.
  - Non Group 3 fax machines.
  - For these devices, database tagging per administration is still required.
  - The "clear channel" (classmarks = 0) is no longer feasible for voice, fax or modem. It is only allowed (and necessary) for digital data (ISDN) links. Using the clear channel erroneously for voice will disable any fax/modem tone detection and therefore has to be avoided.

2.3.7.2 Übertragung der Ziffern per DTMF

DTMF tones are detected by loadware automatically using the in-band mechanism described in RFC 2833.

Using voice compression in an IPDA architecture a reliable in-band transport of DTMF digits only can be guaranteed using the in-band mechanism described in RFC 2833. In IPDA the implemented DTMF detection and transmission works according to RFC 2833.

With the Gateways, tone detection is provided using the DSP modules which recognize a DTMF tone and transmit it in-band to the opposite side without using an additional connection for signaling.

Each Voice channel contains a DTMF detector in Receive- and a DTMF-sender in Transmit direction. In DTMF-Receive direction a DTMF suppressor is provided. To avoid misdetection within normal conversation a voice protection mechanism is provided.

General Controls

– DTMF Detection ON / OFF.
– The DTMF-Receiver is compliant to ITU Q.24.
– The DTMF -Transmitter is compliant to ITU Q.23.

2.3.8 Redundancy (Standby Board)

To reduce the negative effects if the board fails, it is possible to switch over to a standby board HG3500 within the same HiPath 4000 System.

The switch over to a standby board is optional. It can be administrated per active board via AMO if a switch over to a standby board has to be done or not.

The switch over mechanism is performed by a failure of the board itself, by plugging off an active board and by a defective or a plugged off LAN cable. The dependability has to detect a failure.

The respective active and standby boards have to be pooled in order to enable the switch over. The automatic switch over will only be done within the same board reconfiguration pool. In case the pool consists of one or more active boards and only one standby board, a defect of an active board will cause a switch over to this specific standby board. In case the pool consists of one or more active boards and more than one standby board, a defect of an active board will cause a switch over to an arbitrary standby board. An assignment of the standby boards according to the system topology is possible (e.g. assignment to specific (IPDA)-shelves or to Host or mixed).

Standby boards are connected with the LAN. All higher protocol layers are disabled. Therefore no IP address is assigned to the standby board. In case of a switch over, the IP address of the active board is moved to the standby board and the active board becomes a defective standby board without IP address.
All affected active boards must be registered within the same local IP subnet (LIS) with a unique IP address. If the routers after switch over to a standby board don’t get an answer with the old MAC address they start an ARP-Request to receive automatically the new MAC address for the given IP address. The board which takes over the functions, also initiates an immediate ARP request (gratuitous ARP) in order to actualise the address resolution tables in the IP phones, if e.g. the LAN is not equipped with routers.

By configuring a pool attribute it can be controlled, whether a single or a multiple switch over is possible (hop control).

If a switch over has occurred, the Network Management is informed by the new alarm (Per-Board Switchover)

At the user terminal, the switch over is visible by the new "Board Reconfigure" message

Manual switch over

The function of a HG3500 board can be switched over to a standby board by AMO. The selection of the standby board can be done from the administrated pool or by direct choice.

The Configuration Management supports the standby board configuration , that means for the assistant the administration (install, change, cancel) of standby boards, standby board pools and assignment of active boards to standby board pools.

The support of standby board and switch over mechanism does not increase the number of system licenses (AMO CODEW)

Restrictions

– Both boards (active and standby boards) must be identical hardware.

– The board must have been in service (DC state ‘Ready’) before a board error can initiate the switch over. During startup of the board, of the shelf or of the system, the switch over is not performed.

– A failure of a shelf (including IPDA Access Points!) to which a board belongs or a failure of the LAN/WAN between Host switch and the Access Point does not cause a switch over to a standby board.

– Redundancy for single functionalities out of the multiple feature support configuration of a common gateway is not provided.

– For the HG3575 is no switch over to a standby board possible.

– Physically the 2nd LAN interface is available at the HG 3500 board. But this is already in use for WAML, which requires two active LAN interfaces (customer LAN and ATLANTIC LAN). This means, that a board with WAML functionality does not work together with the redundancy feature! If WAML functionality is configured there is no LAN redundancy at all.

2.3.9 Second LAN Interface

The common gateway uses the redundancy Ethernet interface on the board. There are two physical LAN connectors which can be applied to different IP-switch devices for redundancy. If the actually active connection to customer LAN fails (Layer 1 error) the redundant LAN- connection takes over the functionality. The 2nd LAN- interface will have the same MAC and IP address.

Restrictions

Physically the 2nd LAN interface is available at the HG 3500 board. But this is already in use for WAML, which requires two active LAN interfaces (customer LAN and ATLANTIC LAN). This means, that a board with WAML functionality does not work together with the redundancy feature! If WAML functionality is configured there is no LAN redundancy at all.

2.3.10 Large Enterprise Gatekeeper

Two or more HiPath 4000 systems can be connected via an IP-based network using HG 3500 Gateways. In IP-trunking scenarios the HiPath 4000 Large Enterprise Gatekeeper (LEGK) is used to support address resolution, output of IP address and number assignment, closed numbering plan and access to LCR of HiPath 4000 for up to max. 30 registered gateways and max. 500 destination gateways.

In addition, the Resource Manager of the Large Enterprise Gatekeeper provides monitoring of network component utilization and controlling of direct media connections and IPDA traffic.

The basis of the Large Enterprise Gatekeeper (LEGK) is the LCR functionality of HiPath 4000 and the HG 3500 gateway. This routing mechanisms allows an intelligent routing to the desired destination over the IP network.

2.3.10.1 Features

The Large Enterprise Gatekeeper of HiPath 4000 supports the following features:

– Adress resolution

– Resource management

– H.235 Security Oscar Profile 1

Configuration of the Gatewaytable

Each LEGK has a list of known gateways in a so-called gateway table. This list specifies all possible destination gateways and all internal gateways that can register with the respective LEGK. It contains static data for each gateway (e.g. IP address and sector number) as well as dynamic (semi-permanent) data, such as current registration data, etc. This list contains a maximum of 500 entries. The table is addressed using the numbers for the HG3500 gateways.

RAS protocol and address resolution

The H.225 RAS protocol (Admission and Status) is a part of H.323 protocol and it serves for communication between gatekeeper and gateways. The RAS protocol is used for the request and control of status information. The Gateway HG 3500 maps the RAS messages to the corresponding messages, which are implemented in the Cornet-NQ protocol for IP Address Resolution.

Address Resolution

For address resolution the number of the destination endpoint is analyzed and the IP address list is read from LCR data.
Address resolution is provided for E.164-, private numbering plans (PNP) and implicit plans. Networks with open and closed numbering are supported. The address resolution is supported in accordance with H.225 procedures.
In the LCR route tables the IP addresses of the destination gateways are included.

The HG 3500 receives a H.225 request , e.g. with a E.164 number to be resolved. If it is authenticated by the LEGK in accordance with H.235 Security Oscar Profile 1, that the number belongs to a local endpoint, the IP address of the corresponding gateway HG 3500 is returned as result to the Gatekeeper. Otherwise if the number belongs to an external destination, the route and corresponding list of route elements are determined on the basis of the LCR dial plan.
This list is also assigned a list of IP addresses for the corresponding destination gateways. In the analysis of the LCR tables route selection is provided with the normal criteria like classmarks, time of day, etc. If no classmarks were received, the default classmarks from the gateway table are used.
Then the Resource Manager checks if there are sufficient resources (B-channels in
the destination gateway) available for the individual destination gateways. A list of positively checked IP addresses is returned to the Gatekeeper.

2.3.10.2 Ressourcemanager

Resource management

Each HiPath 4000 has a Resource Manager, which as part of the LEGK provides the management of resources (e.g. use of bandwidth for WAN connections) in the network. For this the Resource Manager monitors utilisation of the individual gateways in the network and prevents unnecessary seizure attempts on utilised or defective destination gateways.

The Resource manager is able to manage max. 500 sectors. A sector is a part of the network that provides a specific bandwidth up to 4 x 2 Mbps (4 x 30 B-channels) with one HG 3500 gateway, depending on the type and extension. Each sector is allocated one or more Resource Manager. The Resource manager monitors the resource utilization in the sector and can monitor multiple sectors (up to max. 500).

The following items are important for the resource management of the LEGK:

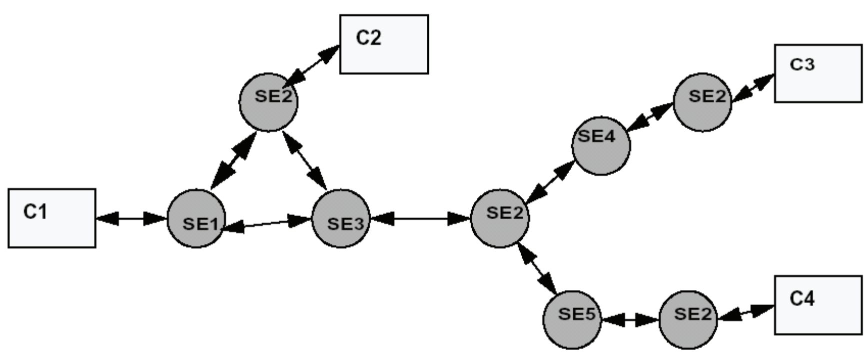
– network topology
– sector management
– network-wide data replication

Network topology

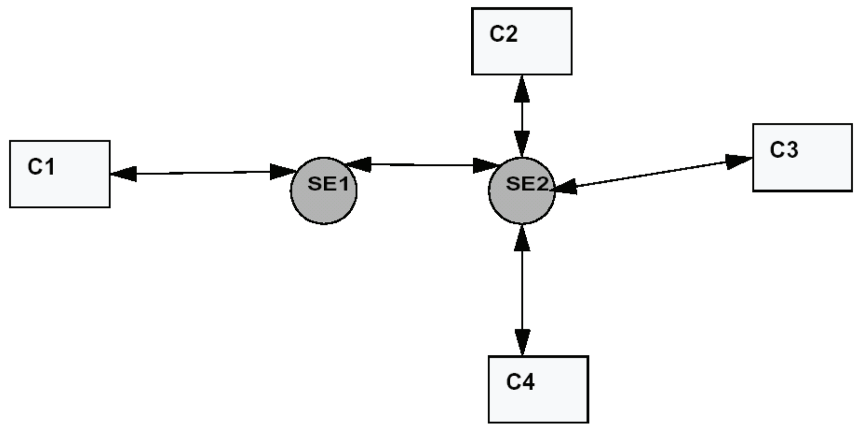
To represent the network topology, the endpoints (in this case the devices) are grouped in clusters. A cluster incorporates endpoints that possess the same properties in terms of accessibility. In the example above, the

– TDM subscriber is in Cluster C1,
– 4711 subscriber is in Cluster C2,
– 3500 subscriber is in Cluster C3 and
– 4200 subscriber is in Cluster C4.

The topology or accessibility can then be represented as a directional graph. The following graph shows the accessibility relationship for the master connections:



Below is the graph for direct media connections:



Sector management

Each Resource Manager basically knows all sectors in the network and can be responsible for the management of some sectors. Call processing checks whether there are sufficient resources available for each connection set-up. The Resource Manager can determine which sectors are to be passed through from the required bandwidth, connection type (DMC or hybrid) and endpoint clusters and check whether there is sufficient bandwidth available. In the case of hybrid connections, resource availability can, depending on administration, either be end-to-end or link-by-link, i.e. to the next Resource Manager.

In a second step, call processing provides the local Resource Manager with information about the required bandwidth, the type of connection (DMC or hybrid) and the endpoint clusters for each connection setup (DMC and hybrid). From this information, the Resource Manager can determine the impacted sectors and update the utilisation data for those sectors for which it is responsible. In the case of hybrid connections, only the sectors to the next Resource Manager are updated.

In sector management, a distinction is made between sectors to which just one Resource Manager is assigned and sectors to which multiple Resource Managers are assigned. If a sector is managed by multiple Resource Managers, each of these Resource Managers must know how many resources each of the other Resource Managers use in order to be able to determine the total utilisation.

The total utilisation is then the sum of the individual utilisation values.

Network-wide data replication

For network-wide data replication all Resource Managers are linked via Cornet NQ signaling. With the service capacity reporting the Resource Managers exchange the utilisation data for their sectors during connection setup.

For sectors, managed exclusively by one Resource Manager, the current utilisation data is transmitted. For a shared sector, i.e. a sector to which multiple Resource Managers are assigned, only the resources required by the Resource Manager are transmitted. Upon receipt of the data, the local Resource Manager tables are updated with the new information. This exchange of information takes place during every connection setup.

H.235 Oscar Profile 1

In the network is one single shared secret for all gateways used.

Remarks and Restrictions

In HiPath 4000 the LEGK is only used for IP trunking. The functionality of the LEGK is an interaction between the HiPath 4000 system software and the HG 3500 board. The gateway can use the address resolution capability of the system software. The LEGK functionality is generally available to all network components through the H.225 RAS protocol.

An registration of remote HG 3500 gateways at the HiPath 4000 LEGK will not be supported.

Capacities of LEGK:

- Registration up to 15 internal gateways plus additional 15 external gateways:
- Max. registered gateways: 30
- Max. destination gateways: 500
- Within the context of gateway resource management, up to 500 sectors can be managed.

The Resource Manager functionality is used by direct media connections (DMCs) and generally for IPDA connections.

2.3.11 Access Point Multicast loading

As the common gateway development has combined the software of all previous gateways (HG 3530, HG 3540, HG 3550 and HG 3570) in a single software package, the size of the software has increased accordingly. This has resulted in a loading time of approximately 15 minutes. The load concept was adapted for this reason. The new load concept is also valid for HG 3575 because the Web Based Management supports from now on HG 3575 as well.

– Loadware can be loaded in the background via WBM during normal operation.

– Activate (decompress and install) the loadware. Activation (loadware decompression and installation) now takes less than five minutes.

– The software (loadware) can be activate immediately, or can scheduled by activating the "Date" button to implement time-controlled activation on a particular day.

2.4 HiPath 4000 Softgate

HiPath 4000 SoftGate is a software application, which includes full HiPath Feature Access for IP-phones and IPDA functionality based on standard server hardware.

The main SW components in the HiPath 4000 SoftGate Application are:

– Integrated MGCP media server generating audio tones, announcements, Music on Hold and providing audio conferencing capabilities.

– Virtual HG 3575 for signaling with HiPath 4000 communication server

– Virtual HG 3500 for connectivity of IP endpoints (HFA and SIP) and SIP-trunking

– Signaling and payload encryption between hostsysytem, virtual HiPath HG3575 and HiPath HG 3570 or other IP Access Points/SoftGates

– Signaling survivability

– Connectivity to AP-emergency server (integrated in IP Access Points)

– Connectivity to Disaster Recovery Server

HiPath 4000 SoftGate offers the following features:

– HiPath 4000 SoftGate 50 for up to 50 IP stations (HFA and SIP stations)

– HiPath 4000 SoftGate 1000 for up to 1000 IP stations (HFA and SIP stations)

Virtual HG 3500

The virtual HG 3500 can be configured as "HFA only" mode or "SIP" mode.

Virtual HG 3500 in HFA Mode provides connectivity for example:

– IP-telephones,

- HFA WLAN Phones
- AC-WIN IP
- OpenScape Personal Edition HFA V3

Virtual HG 3500 in SIP mode enables following SIP connectivities:

- SIP Subscriber,
- Native SIP Trunking with own or 3rd party applications,
- SIP-Q Trunking or
- SIP Sedvice Provider Connectivity

Supported SIP Devices for virtual HG 3500 SIP for example are:

- SIP-telephone
- OpenScape video endpoints
- Mediatrix 41xx for analogue phone connectivity
- Mediatrix 44xx for S0 subscriber connectivity
- DECT over IP base stations
- HiPath MobileConnect devices

For further informations regarding the supported devices please refer to the actual sales release.

2.4.1 Features

HiPath 4000 SoftGate offers the following features:

- Pure software solution
- Runs on standard server hardware (Fujitsu Siemens Computer®, IBM®)
- Operated with the Suse Linux Enterprise (SLES 10 SP2) standard operating system
- Administration via HiPath 4000 Assistant/Manager
- Support of IP- and SIP telephones
- Supports native SIP and SIP-Q trunking
- The SIP service provider connection is integrated in the application. SBC (session border control) configuration is dependent on the customer network. SBC is needed for:
  - network adress translation (NAT)

  - Firewall
  - Hunting
- Supports ISDN calls in the public network via Mediatrix SIP gateway or AP 3700 IP.
- analog and fax connection via Mediatrix 41xx SIP gateway or AP 3700 IP
- S0 connection via Mediatrix 44xx SIP gateways
- integrated media server
- Video integration with OpenScape VHD terminals
- Cordless telephony with DECT over IP
- End-to-end payload connections between native SIP trunks and SIP stations
- Supports HiPath 4000 IPDA survivability options: signaling survivability and connection to existing AP emergency servers
- The number of HiPath 4000 SoftGate 50/1000s or IP access points that can be connected to a HiPath 4000 system is 83.

Support of redundant LAN-Interface (Bonding functionality from LINUX).

- If both LAN cables are connected and the SoftGate Server is starting up, LAN port 1 will always be activated. LAN port 2 will be on standby.
- If only one LAN port is connected when the SoftGate Server is starting up (LAN1 or LAN2), that port will be used.

If the active LAN port is disconnected/disabled by peer or equipment (when both LAN ports are connected):

- The operating system LINUX activates the standby LAN port.
- The "new" active LAN port sends a GRATUITOUS ARP with the same MAC and IP addresses as the "old" port.
- When the operating system LINUX switches ports, the payload will be lost for < 2 sec - all active connections will be saved and NOT disconnected.
- If the "old" port comes up again, no port switchback will be performed.

2.4.2 Restrictions

- No DMC support on virtual NCUI.
- Analog modem connections are not supported.
- No AP1120 support.

- Signaling and payload encryption for stations and SIP trunking are not supported.
- QDC is not supported.
- H.323 IP trunking or CorNet IP trunking are not supported.
- Codec G.723 is not supported.
- Up to 120 parallel connections from vHG3575.

2.4.3 ISDN-Connection on Mediatrix-Gateway

- A Mediatrix 44xx gateway can simultaneously support both an ISDN subscriber line and a line to the private network.
- Every station at a Mediatrix gateway must be configured as an SIP subscriber in the HiPath 4000.
- Mediatrix 44xx gateways do not support telephony features for ISDN phones.
- Refreshing is not supported for ISDN phone displays.
- ISDN data connections are only supported for stations or trunks that are configured on the same HiPath 4000 SoftGate. DMC via LAN is not supported for S0-data connections.
- All gateways should be synchronized with a common clock. For instance, the gateways receive the clock pulse via a direct connection to the public network or via an internal ISDN S0 line (layer-0 connection) to another synchronized gateway.
- Subscriber calls (i.e. for incoming calls from Mediatrix gateway) always have to supply their calling number (internal/extension number). If the ISDN terminal does not do this, calling number insertion can also be configured in the Mediatrix gateway as required.
- A Mediatrix 44xx gateway can support up to 48 network interfaces. A HiPath 4000 SoftGate (or vHG3500) can only operate one network interface in the Mediatrix gateway.
- The Mediatrix gateway’s "Telephony - Call Routing Config" setting is not used in the general scenario for CO trunks (especially not for differentiation based on E.164 number). Station number handling is usually configured in the HiPath system.
- If a Mediatrix gateway is configured for subscribers and CO trunking via the same HiPath 4000 SoftGate (vHG3500), the "Call Routing" must be configured in the Mediatrix gateway.

2.4.4 Analog Station on Mediatrix-Gateway

- Analog stations for the "Voice" or "Fax" service can be configured at a Mediatrix 41xx gateway.
- Analog stations in a Mediatrix 41xx gateway must be configured as SIP subscribers in HiPath 4000.

2.4.5 Video Connection

HiPath 4000 SoftGate offers video support for internal SIP video endpoints. The SIP video endpoints can be connected directly to a HiPath 4000 or to networked HiPath 4000 systems. The functionality is divided in:

- Connection between video endpoints
- Connection between video endpoints and audio endpoints

Prerequisites:

- The system treats SIP video endpoints as normal SIP subscribers.
- A video connection is only possible when a DMC connection is already ongoing.
- All video endpoints (OpenScape VHD) must support DMC call flows.

Connection between vvdeo endpoints

- Peer-to-peer Video communication
- 3- party video conferencer
- Support for other features (such as CLIP, CLIR, COLP, COLR) depends on the video endpoint used

Connection between video endpoints and audio endpoints

Active feature support for video endpoints -
features are initiated by the video endpoint for the audio endpoint.

- regarding 3-party conference with audio and video endpoints
  - An audio endpoint is added to an existing video connection between two video endpoints => the video connection between the two video endpoints stays alive.
  - A video endpoint is added to an existing voice connection between an audio and a video endpoint => a video connection is set up between the two video endpoints

Passive feature support for video endpoints -
features are initiated by the audio-only endpoint for the video endpoint. This could be

- Basic audio call
- Local SIP 3-party conference
- All SIP features enabled for the video endpoints used

Restrictions:

- No interworking with HFA video endpoints and TDM video endpoints.
- Video is only possible for DMC slave connections.
- The DMC connection is interrupted if a feature such as "hold" or "transfer" is used. Video connections cannot be established while these functions are active. The DMC connection is only re-established and video connections can only be set up when these features have been deactivated.

2.4.6 DECT over IP

- Coming soon. Function will be part of a later release.

2.4.7 Comparison of functions of HiPath SoftGate vs. AP3700 IP

Content	HiPath 4000 SoftGate with virtual HG 3500	Access Point AP 3700 IP with Gateway HG 3500
Hardware	Predefined FSC and IBM Server	HiPath HG 3575 (NCUI4) HIPath HG 3500 (STMI4)
Operating System	SUSE Linux (SLES 10 Sp2)	VxWorks
LAN Speed	Predefined servers support up to 1 Gbit Ethernet Controller	10/100Mbits Full Duplex
Interfaces		
HFA-Phones	Yes	Yes
SIP-Phones	Yes	Yes
SIP-Qv1 Trunking	Yes	Yes
SIP-Qv2 Trunking	Yes	Yes
Native SIP Trunking	Yes	Yes
SIP Service Provider	Yes	Yes

Content	HiPath 4000 SoftGate with virtual HG 3500	Access Point AP 3700 IP with Gateway HG 3500
Cornet IP Trunking	No	Yes
Codecs		
G.711	Yes, Sample Size: 10-60ms	Yes, Sample Size: 10-60ms
G.729	Yes, Sample Size: 20-80ms	Yes, Sample Size: 20-80ms
G.723	No	Yes
Voice Activity Detection	Yes	Yes
Comfort Noise Generation	Yes	Yes
T.38 Fax Connections	T.38 Fax Connections cannot be terminated by virtual HG 3500. T38 Fax only possible, when DMC is activated for the used SIP Subscriber and native SIP Trunks	Yes for SIP Subscriber Native SIP Trunking SIP Service Provider Cornet IP Trunking
Resilience Features		
Redundant LAN Ports	Yes	Yes
Signaling Survivability	Yes	Yes
Payload Survivability	Yes, but only between Hostsystem and Softgate	Yes
Support of AP-Emergency Server	Yes, but AP-E cannot be integrated	Yes (AP-E can be integrated)
HFA Stand-By	No	Yes
Disaster Recovery	Yes	Yes
QoS Functionality		
• L2 QoS IEEE 802.1 p/q (VLAN Tagging) on Layer 2	Yes	Yes
• L3 QoS conform to IETF RFC 2474 (DiffServ)	Yes	Yes
QoS Data Collection	No	Yes
Security		
Signaling and Payload Encryption for HFA Endpoints	No	Yes
Signaling and Payload Encryption for SIP Endpoints	No	No
Signaling and Payload Encryption for SIP-Q Trunking	No	Yes

Content	HiPath 4000 SoftGate with virtual HG 3500	Access Point AP 3700 IP with Gateway HG 3500
Signaling and Payload Encryption for IPDA/ SoftGate Links	Yes	Yes
Common Gateway Features		
Number Common Gateways	Maximum 9 virtual Common Gateways can be configured per installed SoftGate Server	Maximum 6 Common Gateways in AP3700 IP and 10 in AP3700
Common Gateway Standby Card	No	Yes
Maximum Users	Max 240 per virtual HG 3500	Max 240 per HG 3500
Maximum Channels	Max 120 per virtual HG 3500	Depending on used STMI4 Hardware and activated functionalities (SPE, QDC)
End To End Payload between IP Endpoints	Yes	Yes (DMC)
SNMP Network Management	Yes	Yes
HFA Feature		
HiPath 4000 ComScendo Feature Set	Yes	Yes
SIP Feature		
SIP Subscriber features	Yes	Yes
MWI for native SIP Trunking	Yes	Yes
Name Display for native SIP Trunking	Yes	Yes
Call Transfer for native SIP Trunking	Yes	Yes
Video Support for SIP Subscriber	Yes	No
Common Gateway General		
End To End Payload between IP endpoints (unsing DMC)	Yes	Yes
SNMP Network Management	Yes	Yes
WAML Replacement (PPP)	No	Yes

2.5 HiPath Payload Switching

By the use of the feature "DMC Any-to-any" payload data is transported within a HiPath 4000 network direct between the IP Endpoints without several IP-TDM conversions of the payload. This direct payload connection is called Direct Media Connection (DMC).

The IP endpoints can be the real endpoints of the connection (i.e. there is a Master Connection between 2 IP phones) or the IP endpoints are only endpoints of a section within the Master Connection. E.g. an Anate or Digite may also profit from a DMC if it is connected to its partner via IPDA Access Point or IP Trunking Gateway.

The setup of a call is done within several steps.
Steps 1 and 2 are the normal call setup procedures as known in HiPath 4000 V1.0. The result is called Master Connection. The normal call setup procedure is extended by steps 3 and 4 to build the optional Direct Media Connection:

- Step 1: In a first step, a signaling connection between the users is established.
- Step 2: Parallel to that signaling connection a so called Master Connection (MC) for the payload is established. The payload path of the Master Connection may be segmented into TDM and IP paths:
  - within a TDM network this is a specific B channel
  - within an IP network, this is a RTP connection.

  The RTP payload connection is established and controlled by H.323 procedures (in particular by the use of the fast connect procedure).
- Step 3: After the Master Connection between the users is established and connected, i.e. both users are in the talk-state, a DMC connection between the IP endpoints of the Master Connection can be established. The called IP endpoint initiates the setup of the DMC connection.
- Step 4: If the DMC connection is established, the payload is transported using this H.323 connection. However, the Master Connection exists in parallel, but without conveying payload as long as the DMC connection exists (the Master Connection is on a "stand-by" mode).
  If the talk-state is released (e.g. for the setup of a consultation call), the IP endpoint switches back immediately to the Master Connection and releases the DMC connection.

Restrictions

The Direct Media Connection (DMC) will be switched only in a two party call. DMC will also be switched, if the call is setup via a feature like call forwarding or picking up a group call or if a secondary line answers a keyset call.

If the involved parties aren't in a 2 party connection, the IP/TDM conversion and the switching via the TDM switching network (MTS) is executed.

A DMC is not switched for conference calls.

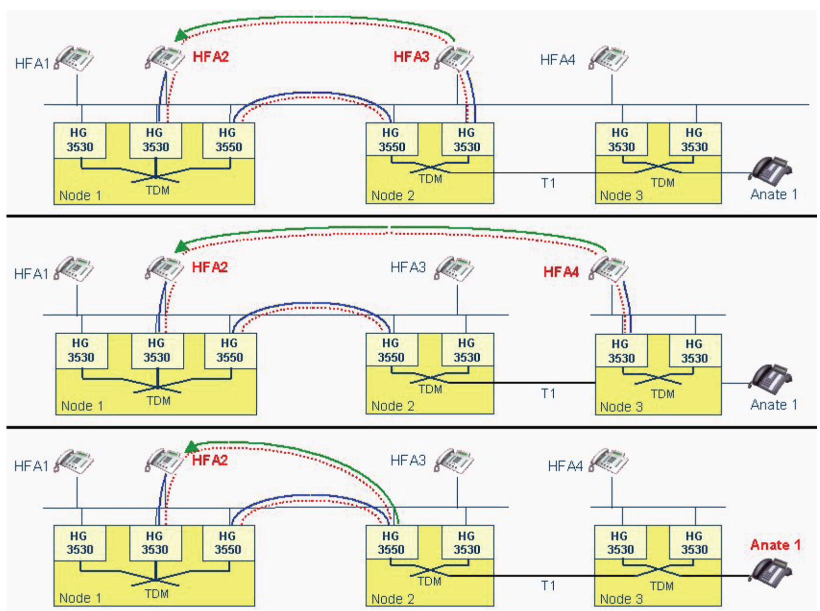
If an Direct Media Connection is established, the connection path via the MTS switching network for multiple IP/TDM conversion (Master Connection) has to stay switched in parallel to the direct IP connection so that immediately after termination of the 2 party call state the MTS with its capabilities for tones and conferencing is available for any feature control by call processing.

DMCs are established for voice and Fax/Modem connections.

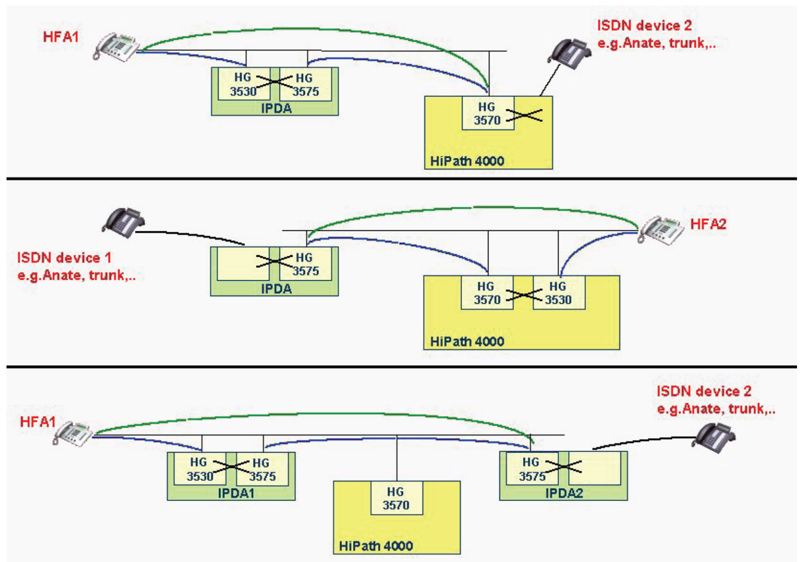
2.5.1 Scenarios

The following scenarios are been considered:

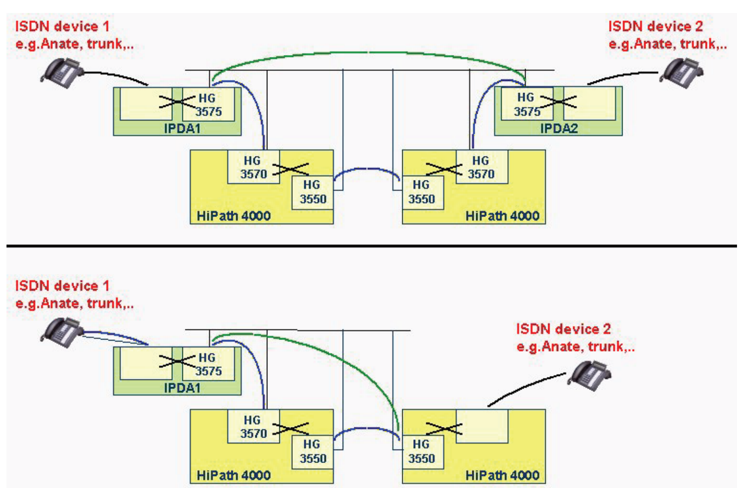
Scenario A: Direct Media Connections DMC between HFA IP telephones within a HiPath 4000 network.



Scenario B: Direct Media Connections DMC between HFA IP phone and IPDA shelf within a single HiPath 4000 node.



Scenario C: Direct Media Connections DMC between netwide IPDA-shelves.



DMC Interworking to HiPath 3000

Supported scenarios:

– Scenario 1: HG 3500 terminates the HiPath 3000 payload switching

– Scenario 2: HG 1500 terminates HiPath 4000 Direct-Media-Connection

– Scenario 3: Direct-Media-Connection between HFA IP phones end-to-end

– Scenario 4: HiPath 4000 as transit node between two HiPath 3000 nodes

Basics of the solution:

– DMC with HiPath 4000 and

– Payload Switching (PLS) in HiPath 3000

– i.e. HG 3500 of HiPath 4000 controls the HiPath 3000 PLS

Remarks and Restrictions

– DMC connections with an IPDA board or IP trunking as originating or terminating endpoint will exist, if the endpoint of the basic connection at the IPDA shelf is an analog telephone, a digital systemtelephone, a CMI Cordless phone or an analogue or digital trunk, but not for attendant console.

Feature response in exception situations

– DTMF Handling
A DMC is released, if the endpoint requests the sending of DTMF via SIU (signaling unit). E.g. a IP telephone requests the sending of DTMF via access code.

– DTMF transmission over DMC
DTMF tones are recognized in DMCs and transmitted, e.g. if the tone is received by the HiPath system at an CO interface or an DTMF enable Anate sends DTMF by itself.

2.6 PKI Integration for signaling and payload encryption

2.6.1 PKI-Integration - certificate generation and deployment

The following functions are implemented via DLS server:

- import of certificates generated by a customer PKI (import interface only)
- distribution of certificates
- import and distribution of certificate revocation lists (CRLs) for VPN/IPsec (HiPath 3000 only)
- distribution of CRL distribution points
- distribution of security profile parameters

- generation and distribution of certificates – without CRL support - used to secure the communication path between DLS and DLS Client
- generation of certificates without CRL support needed by the feature “Signaling and Payload Encryption.

There is no single tool for generation of all needed public-key-pairs/certificates to those customers, who do not run an own PKI. Instead, they must use different certificate generation tools for different purposes. These Tools are:

– certificate generator for DLS <-> DLS Client traffic on DLS,

– certificate generator for WBM on CGW,

– certificate generator for SPE entities on DLS

There is no direct interaction (by means of a RA/CA adapter) between the PKI solution and the customer PKI. Instead, certificates and private keys created by a customer PKI are imported into DLS via PKCS#12, PKCS#7 or PEM files and appropriate import instructions.

As far as gateways are concerned, import of certificates and private keys is supported for the following certificate types:

- Peer certificates/private keys for SPE (one per gateway)
- CA certificates for SPE (1 to 16)
- Peer certificates/private keys for IPSec/VPN (1 to 16 per gateway)
- CA certificates and CRLs for IPSec/VPN (1 to 16)
- Peer certificates for WBM server (1 to 16 per gateway)

2.6.1.1 Certificates for DLS interaction protocol

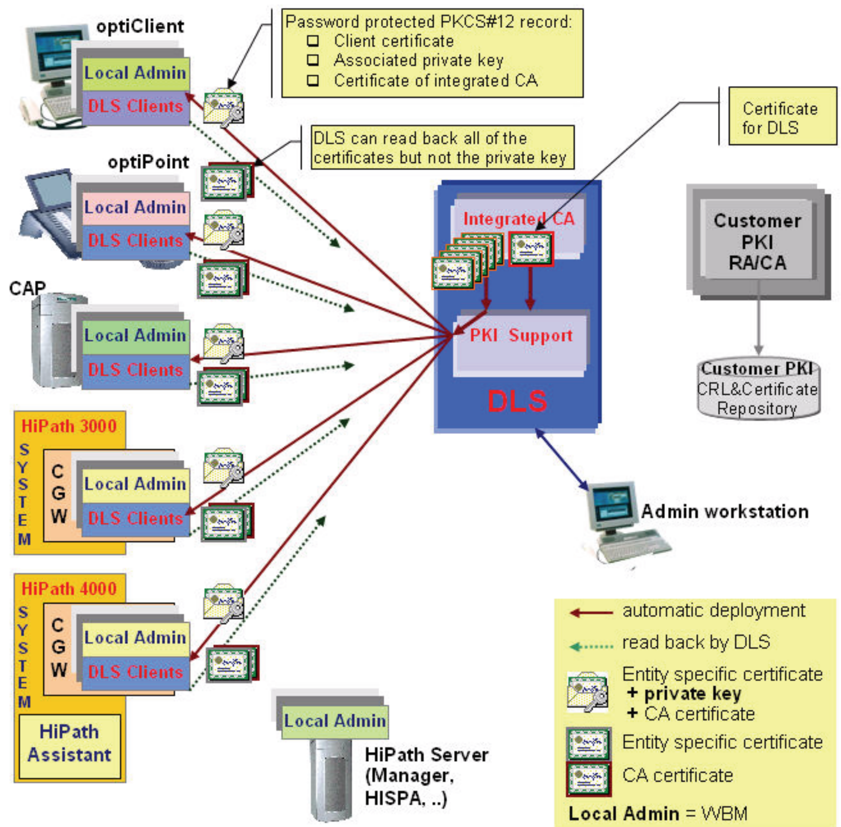


Figure 12 *Certificate generation and deployment for DLS / DLS Client communication*

All certificates and private keys used for securing the communication path DLS to DLS Client are issued by a self-signed CA on DLS and deployed by DLS exclusively. Therefore no import of certificates and private keys from a customer PKI or via local admin at the DLS Clients are possible.

Deployment of individual DLS certificates and private keys is a prerequisite for any other DLS interaction. Therefore, this deployment is done in a special manner, called "DLS Client bootstrapping".

2.6.1.2 Certificates for secure WBM access

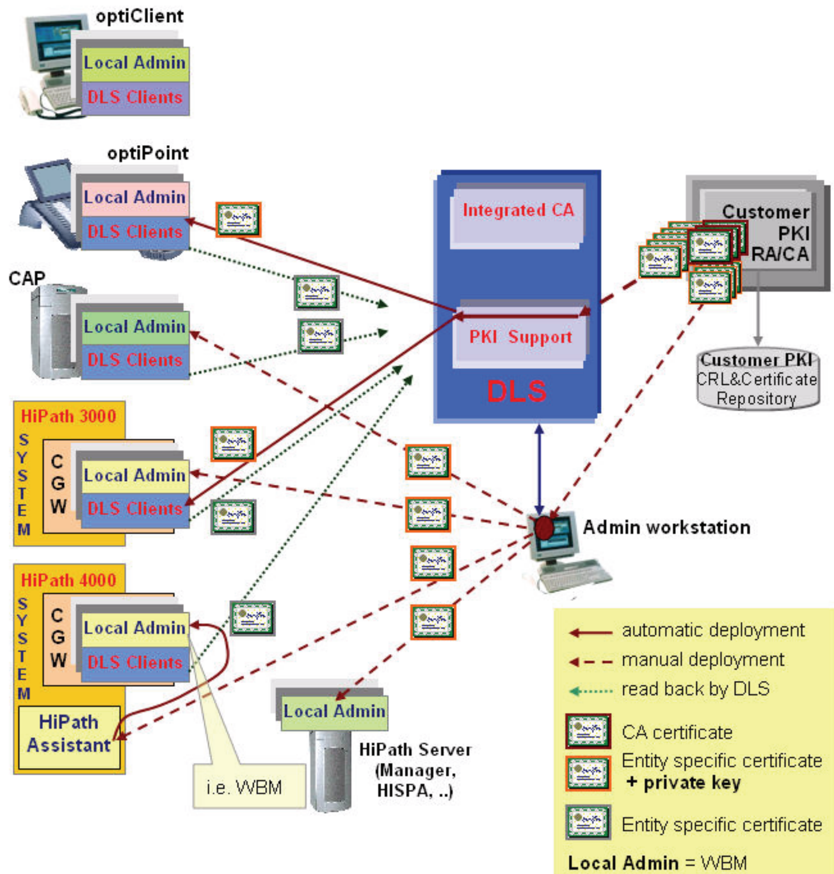


Figure 13 Certificate generation and deployment for secure WBM access

As shown by Figure above, the DLS supports the import and distribution of certificates needed for securing the WBM access of IP-Telephones and Common Gateways belonging to an HiPath 3000 system.

This functionality is **not available** for IP-Telephones and CGWs belonging to an HiPath 4000 system. Note:

- optiClient has no WBM at all. Configuration is done via Windows dialogs.
- Because of the implemented concept for CGW administration through the HiPath Assistant, all common gatways in a HiPath 4000 need the same certificate and key material as the HiPath Assistant itself. Thus, the HiPath Assistant streams down its own WBM certificate and key material to all gateways and the gateway must not accept any WBM certificates received from the DLS.

Because the WBM certificates are used for securing the WEB server only and mutual authentication is not required for WBM access there is no support for CA certificates and CRL needed.

If no customer PKI is available, one of the already existing CA implementations may be used instead. Thus, in an HiPath 4000 environment the so called PKI-light on the HiPath Assistant (Note: Certificate generation for SSL is not available on Common gateway in HiPath 4000) may be used to generate the WBM certificates. .

Regarding CRL - that may be used by the WEB Clients when accessing an HiPath entity - it has to be mentioned that:

- PKI-light on the HiPath Assistant has not implemented any CRL support at all. Thus, if the PKI-light is used a CRL will not be available at all.
- Certificate generation for SSL does not support CRL.

2.6.2 Certificates for Signaling and Payload Encryption

For signaling and payload encryption, the scenario "Certificate generation and deployment for SPE from customerPKI" is supported.

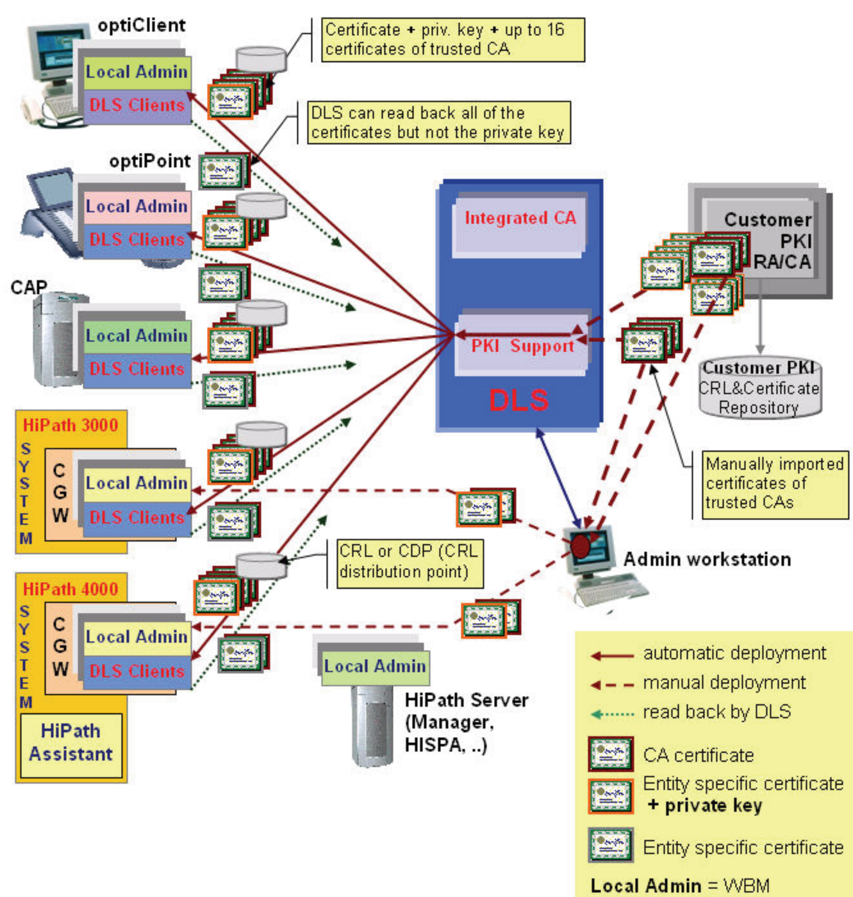


Figure 14 Certificate generation and deployment for SPE: customerPKI

Scenario:

- All certificates and private keys are issued by a customer PKI
- DLS supports the import and distribution of those certificates and private keys
- DLS supports distribution of trusted CA certificates
- DLS supports the distribution of the CRL distribution point
- DLS supports the distribution of the Security Profile parameters
- Common Gatway supports the import of all SPE relevant PKI data via WBM too

For signaling and payload encryption, the scenario "Certificate generation and deployment for SPE - DLS integrated CA" is supported.

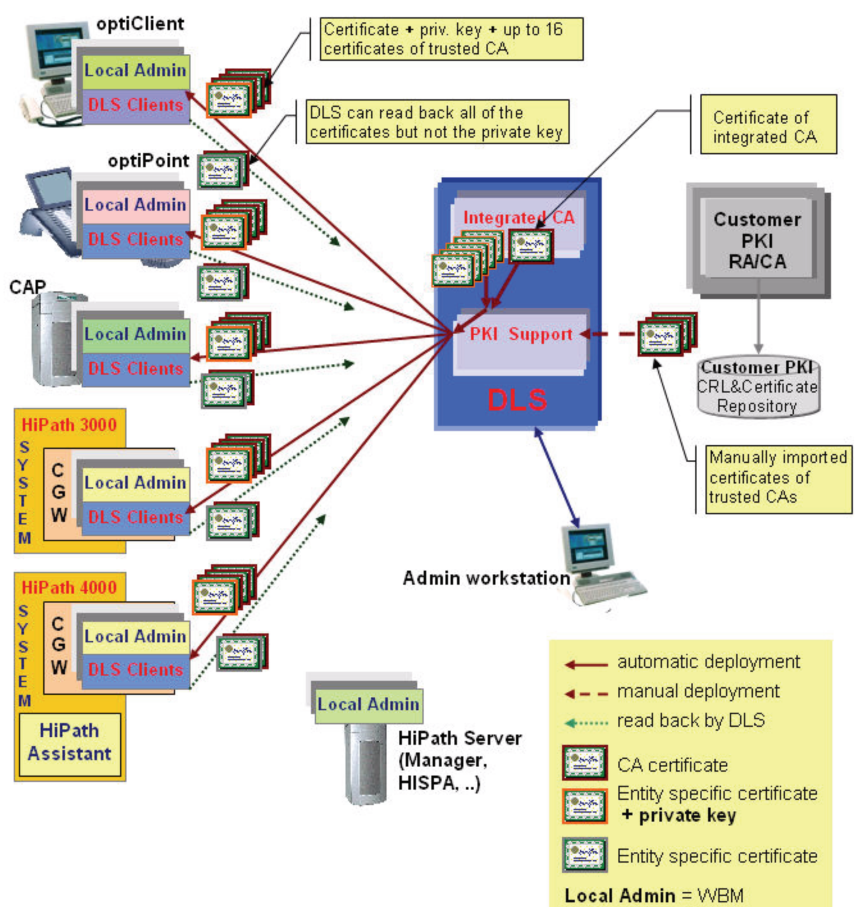


Figure 15 Certificate generation and deployment for SPE: DLS integrated CA

Scenario :

- All certificates and private keys are issued by the integrated CA of DLS, also called PKG (Public Key Generator)

- DLS supports the distribution of those certificates and private keys via the DLS interaction protocol exclusively
- DLS supports the distribution of trusted CA certificates
- There is no CRL support at all for this DLS integrated CA - however distribution of CRL distribution points is supported for manually imported trusted CA certificates.
- DLS supports the distribution of the Security Profile parameters

2.7 Signaling and Payload Encryption (SPE)

2.7.1 Signaling and Payload Streams

The distributed and open system architecture of the IP network itself enables attacks against integrity and confidentiality of data communicated over it. The following figure gives an overview of the signaling and payload streams of HiPath systems on the IP network.

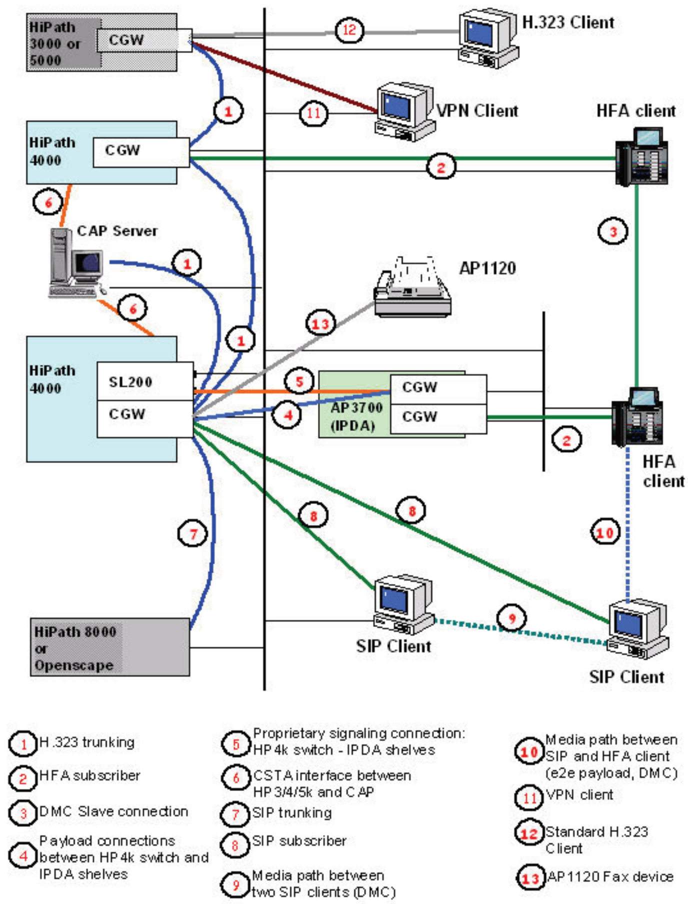


Figure 16 Signaling and payload streams

2.7.2 Features

The function of signaling and payload encryption implement a security system for IP based telephony communication within an HP3/4/5k landscape. This applies to all IP based communication between

- HiPath node <-> HiPath node
- HiPath node <-> VoIP clients
- VoIP clients <-> VoIP clients

The implementation based on a set of functional blocks. These functional blocks are:

1. Licensing, configuration, administration and maintenance
2. handling the different signaling streams within HiPath scenarios based on PEP, TLS, resp. H.235 Annex D

3. Handlich of the payload streams within HiPath scenarios based on MIKEY and SRTP belonging to an HFA client call including DMC.
4. extensions of CorNet-IP/TC, CorNet- IP/TS and H.225 CS respectively H.235 Annex G are implemented by the IP-Clients (HFA) accordingly.
5. Proprietary Solution for the payload streams between HiPath 4000 and HG 3575
6. Involved connections are:
   - Signaling and payload encryption of the HFA subscriber interface
   - Signaling and payload encryption of the H.323 Trunking interface
   - Signaling protection and payload encryption of DMC slave calls
   - Signaling and payload encryption of the SIP trunking interface
   - Signaling and payload encryption between the HiPath 4000 and the HG 3575
   - Encryption of the CTI interface used by CAP
7. all routing and classification whether the end-to-end communication path between the subscribers is secure or not. The following connections are available in an HiPath network:
   - PSTN connections to an other node
   - TDM connections to an other node
   - Secured IP connections to an other node
   - Unsecured IP connections to an other node
8. The following display modifications are relevant ONLY for Secure and Cipher clients in case the signaling and payload encryption feature is activated. There are
   - Check of Security Configuration in Menu (GUI)
   - Check of Security Configuration with Key
   - Displaying of the security level of the active call
   - Notification about the activated security level when no call is active
   - Display after activating Mobile HFA logon (either via access code or menu)
     - Display if the path between visited and home stations is secure or not secure
     - Display if execution of the Mobile HFA feature is not possible
   - Tone Signaling

2.7.3 Definition of security levels

2.7.3.1 Type of VoIP clients

From the user point of view the following types of VoIP clients have to be considered:

- **TRADITIONAL CLIENT:** Today's VoIP clients.

  the term TRADITIONAL CLIENT is used for VoIP Clients without Signaling Encryption. In other cases, .

- **STANDARD CLIENT:**
  A VoIP client that is not able or not allowed (by means of licensing) to encrypt the payload streams is called a standard client, regardless if the signaling connection is unsecured, protected via H.235 Annex D or even secured via TLS.

- **SECURE CLIENT:**
  A VoIP client is called a SECURE CLIENT if the signaling connections between the client and its CGW are secured via TLS and in addition the client is able and configured to encrypt the payload streams.

  Each call originated by a SECURE CLIENT is requested as an interception safe call. That means, that the HiPath system is requested to provide encryption of the signaling and payload connection of all network links between the called and calling party, if possible. If that cannot be assured, the HiPath informs the user and establishes an unencrypted call.

- **CIPHER CLIENT HiPath 4000 & HFA devices ONLY:**
  From the user point of view this client is identical to a SECURE CLIENT except that "Cipher Client" instead of "Secure Client" is displayed as security configuration status.
  The particularity of a CIPHER CLIENT is that it sends and receives encrypted payload only regardless of the end-to-end security level of the call. This will to be assured by the HiPath switch in the way, that a CIPHER CLIENT will never be an endpoint of an unencrypted DMC call.
  Instead, if the other endpoint is a standard client or a network link in between does not support payload encryption, the DMC connection will be established between the other endpoint and a CGW of the switch the CIPHER CLIENT belongs to.

  The significant advantage of this concept is that calls of a CIPHER CLIENT cannot be traced by monitoring the CIPHER CLIENT's device.

2.7.3.2 Type of IP based Trunks

IP based trunks could be differentiated in:

- **TRADITIONAL TRUNK:**Today IP Trunks.
  The term TRADITIONAL TRUNK is used for IP Trunks without Signaling Encryption.
- **STANDARD TRUNK:**
  Payload encryption is not supported on this trunk. Whether the signaling data are encrypted or not depends on the SPE capabilities of both of the involved common gateways. The term STANDARD TRUNK is also used to summarize all VoIP Clients without Payload Encryption.
- **SECURE TRUNK:**
  All signaling and payload data transferred over this link will be encrypted regardless of the requested call security level. A trunk becomes a SECURE Trunk only in case both of the involved common gateways have signaling encryption activated and on both sides payload encryption has been licensed and configured for this link.
- **EXTSECURED TRUNK - HiPath 4000 ONLY:**
  If the IP link between two common gateways is secured by external equipments (e.g. VPN tunnel) the involved getways will classify that trunk like a SECURE Trunk but without providing any security mechanism neither for signaling nor payload connections by its own. the IP link has to be configured as an ExtSecured trunk on both sides.

2.7.3.3 Type of security levels

- **Unsecured:**

  No security mechanism implemented for the signaling data and the payload streams
- **Protected**:

  Signaling data streams are protected via H.235 Baseline Security profile whereas the payload streams remain unprotected
- **Secured:**

  Signaling data streams are protected by the mechanism defined by this feature (e.g. TLS) whereas the payload streams remain unprotected
- Interception safe:

  Signaling data and payload streams are protected by the mechanism defined by this feature. This security level is negotiated on a per-call base if the corresponding signaling connection is secured.

  (interception safe = secured signaling connection + secured payload streams)

Whereas the terms listed above specify the security level the following terms refer to the user point of view with respect to the confidentiality of the data.

- Standard:

  summarize the software security levels unsecured, protected and secured
- Secure:

  is equivalent to the software security level interception safe

The following table gives an overview, which security levels for each type of VoIP clients respectively HiPath nodes are supported.

	Unsecured	Protected	Secured	Interception safe
HFA /H.323 client (IP-telephone)	X	X	X	X
HiPath H.323 node (CorNet IP NQ)	X	X	X	X
HiPath SIP node	X		X	X

2.7.4 Capabilities of VoIP-Clients

The IP- telephones which are associated to these types of users have the following capabilities:

A **standard** client supports:

- Encryption of the signaling connections (CorNet-TC/TS and H.323) between phone and common gateway (via TLS)
- Payload encryption between HFA IP phone and its home common gateway is not supported;
- Outgoing and incoming DMC signaling connections could be authenticated using H.235 Annex D procedures, if the corresponding partner DMC endpoint supports H.235 Annex D too;
- The payload of an outgoing and incoming DMC call is not encrypted.

A **secure** client supports:

- Encryption of the signaling connections (CorNet-TC/TS and H.323) between phone and common gateway (via TLS);
- Payload encryption between HFA IP phone and its home common gateway using the MIKEY mechanism (i.e. MIKEY #0 procedure);
- Outgoing and incoming DMC signaling connections are authenticated using H.235 Annex D procedures, if the corresponding partner DMC endpoint supports H.235 Annex D too;

- If requested by the controlling PBX or by the DMC partner endpoint, the payload of the outgoing or incoming DMC call (controlled by the DMC signaling connection) is encrypted using MIKEY procedures (MIKEY #1). The encryption of the payload in the incoming or outgoing DMC call is indicated by CorNet-TS commands.

A **cipher** client supports:

- Encryption of the signaling connections (CorNet-TC/TS and H.323) between phone and its home common gateway (via TLS);
- Payload encryption between HFA IP phone and its home common gateway using the MIKEY mechanism (i.e. MIKEY#0 procedure);
- Outgoing and incoming DMC signaling connections must be authenticated using H.235 Annex D procedures. If the corresponding DMC endpoint does not support H.235 Annex D, the DMC call is rejected;
- the payload of the outgoing or incoming DMC call (controlled by the DMC signaling connection) must be encrypted using MIKEY procedures (MIKEY#1 procedure).

2.7.5 CorNet-IP/TS enhancements concerning DMC

Because DMC connections do not use an underlying TLS layer, the following definition of the security status of DMC connections applies:

- DMC-traditional

  The H.225.0 signaling connection for the DMC connection is neither encrypted nor protected by an authentication token. Payload is transported unencrypted via RTP.

- DMC-standard

  An H.235 Annex D token is used for message authentication and message integrity checks of H.225.0 messages of the DMC- connection. The H.235 Annex D token is used according the H.235 Annex D recommendation and is based on a preshared secret (PSS-DMC) previously exchanged via the Master Connection. The PSS-DMC is generated by the called HiPath 4000. Payload is transported unencrypted via RTP.

- DMC-secure

  The H.225.0 signaling connection of the DMC connection uses an H.235 Annex D token for authentication and message integrity check of the H.225.0 messages. The H.235 Annex D token is used according the H.235 Annex D recommendation and is based on a preshared secret (PSS-DMC) previously exchanged via the Master Connection. The PSS-DMC is generated by the called HiPath 4000. Payload is encrypted and transported using the SRTP protocol. The key material for SRTP is provided by the DMC endpoint which sets up the DMC connection. The transport of the SRTP key material (TGK) is done via the H.225.0 signaling connection using the MIKEY#1 procedure. If MIKEY#1 has been chosen, the SRTP key material is encrypted and authenticated on base of an additional preshared secret (PSS-TGK) previously exchanged via the Master Connection.

2.7.6 Performance

Encryption of signalling data and payload streams has a notable impact on performance and causes higher bandwidth needs. As a consequence, the following reductions have to be considered:

- The maximum number of subscriber respectively trunking channels supported by a single common gateway is reduced by about 20 - 30%.
- In case of HiPath 4000 and DSCXL as SWU processor card in a MONO switch, BHCA for subscriber of IPDA as well as the maximum number of supported subscriber in IPDA shelves will be reduced by about 10%.
- In case of HiPath 4000 and DSCXL as SWU processor card in a DUAL switch, BHCA for subscriber of IPDA will be reduced by about 8%. No reduction of supported subscriber is expected for this configuration.

2.7.7 Restrictions

- Because of "Signalling and payload encryption" is based digital certificates and these certificates are deployed with help of DLS, these features consider only VoIP clients having a DLS interface implemented
- Each common gateway needs an own certificate plus private key and associated CA certificates in order to support signaling encryption based on TLS as well as the MIKEY Option #3 for SRTP key negotiation. All of this certificates and related configuration data, also called PKI data, are distributed by DLS and either imported on the common gateway via the DLS client or the WBM.
- No security measures for securing T.38 Fax are supported.
- There is no SPE support for native H.323 Clients
- IP-telephones without display are not supported.
- Kerberos V5 are not support
- Compatibility to the security solution implemented in SIPSEC Step 1 (HiPath 4000 V3.0, HiPath 3000 6.0) is not provided.

- In order to make sure that the permanent TLS connection towards the HiPath 8000 is established in time this TLS connection is always initiated by the common gateway. The HiPath 4000 support the dynamic SIP Registration towards the HiPath 8000 only. The static SIP configuration implemented by the HiPath 4000 V3.0 as alternative solution to the dynamic SIP Registration will NOT be available in HiPath 4000 V4.
- Because the SubjectName field of certificates issued for the HiPath 8000 may not contain the IP address and the common gateway in HiPath 4000 does not support DNS, the common gateway cannot proof identity of the communication partner by means of received server certificate.
- The common gateway will not perform any identity checks when establishing a TLS connection towards an external SIP Registrar, even if requested by means of configuration data.

2.8 RG 8300

The RG 8300 is a SIP Gateway for the OpenScape Voice. It enables IP connections with SIP-Q to OpenScape Voice and ISDN T1 or E1 PRI connections to Hicom 300 systems or PSTN.

It is based on the hardware and software of HiPath 4000. The SIP-Q link is realized with the IP Gateway HG 3500 and the link to Hicom 300 / HiPath 4000 is realized with the DIUT2 board. The hardware is the AP 3700 IP with the integrated Survivability Box (APE).

Size options and features

- Hardware size options are:
  – RG 8302 - Two T1 / E1 Ports (1 x AP 3700 IP)
  – RG 8304 - Four T1 / E1 Ports (1 x AP 3700 IP)
  – RG 8308 - Eight T1 / E1 Ports (2 x AP 3700 IP)
- Features for whole RG 8300 family :
  – Supported protocol: Q-SIG, CorNet N, CorNet NQ, DSS-1, CAS
  – SIP-Q V2 to OpenScape Voice V3.1 R2 (or higher)
  – all HiPath 4000 V4 Features for the above listed trunk protocols
  – Integration into existing management systems with SNMP and comfortable configuration with the integrated HiPath Assistant.
- Survivable Media Gateway (SMG):

– The RG 8300 can be used as SMG together with a Comdasys Proxy for the OpenScape Voice. In normal operation the SIP phones are connected to the OpenScape Voice and the Comdasys Proxy forwards the SIP signaling transparent (SIP-Q between OpenScape Voice and RG 8300). In SMG mode the SIP registers on the Comdasys Box (e.g. Convergence 3600) and the RG 8300 is then used for the Hicom 300 or PSTN connections (native SIP between Comdasys and RG 8300).

Connectivity and Compatible Products

- Hicom 300, Hicom 300E, Hicom 300H, HiPath 4000,
- SIP- connectivity to OpenScape Voice
- Connectivity to the local PSTN (Public Switched Telephone Network)
- Meets all CE and UL requirements

2.9 Hardware Configuration

2.9.1 Set up in conventional technology

HiPath 4000		
Basic cabinet	up to 3 shelves	up to 1152 ports
1st expansion cabinet	plus 4 shelves	plus 1536 ports
2nd expansion cabinet	plus 4 shelves	plus 1536 ports
3rd expansion cabinet	plus 4 shelves	plus 1536 ports
		max. 5760 ports

2.9.2 Traffic Intensities based of TDM-Ports

The central control and switching components must be calculated to allow for the traffic load caused by all existing traffic types and traffic devices for all communication services during the peak traffic hour.

A performance value of 1 Erl per data connection must be reached for data terminal traffic. The overall traffic capacity in Erl for the existing or possible dimensioning of the control and switching components must be stated. The values must be valid with a loss of B=0,1 %.

The value for the allowed seizure load is the criterion for the capability of the control and must be stated in BHCA (Busy Hour Call Attempts) as per ITU-T recommendation Q.514.

Static traffic capacity

HiPath variant	Static traffic capacity determined on the basis of		
	timeslot allocation		Switching network
	worst case	best case	
HiPath 4500E	2400 (2790) Erl	3840 Erl	3840 Erl
Loss	0,01 %	0 %	0 %

Table 2 Static traffic capacity

Note: Values in brackets indicate only equipment with new boards.

Dynamic traffic capacity according to ITU-Q-514

HiPath-expansion stage (Architecture – Type, see [1])	Hardwareconfiguration	Maximum Score1
HiPath 4000 cPCI – 01/02/03	DSCXL	93.000
HiPath 4000 cPCI – 05/06	DSCXL	275.500

Important: Regarding the HiPath 4000 platform, the 24-port boards and the two-channel interfaces UP0/E and S0 requires that the following definitions must be carefully considered. Here, a port is a 2-wire interface for the connection of lines or stations (in contrast to other approaches where one B channel equals 1 port).

An analog station seizes 1 port . An S0 station seizes 2 ports (4-wire interface!) and uses 2 B channels. A UP0/E station each seize 1 port and use 2 B channels. For time slot allocation in LTUW a B channel equals one traffic source.

If more than 128 B channels are connected per LTUW half in a HiPath 4000 the traffic values must be derived individually!

The HiPath 4000 has 16 slots, each with 24 ports per LTUW. One LTUS half with 8 slots and 192 ports has access to 128 time slots in the switching network.

If no more than 128 ports per LTUW-half are seized, the B channels are allocated to the time slots without any loss. 1 Erl per port or B channel is achieved.

2.9.3 DC Functional Range

DC functional range for station lines:

- Analog: 2 x 750 ohms + telephone
  - An additional increase in the range of 2 x 750 ohms can be achieved by using a remote extension box (with its own power supply). The ranges for ringing and DTMF dialling must be considered separately.
- Digital with the following interfaces

– S_0: Point-to-point connection <= 1 km depending on cable type
Multipoint connection <= 100 m depending on cable type

– U_{P0E}: depends on the cable and telephone type, normally max.1 km. The maximum possible ranges for selected (standard-) configurations are listed in the PN TI.

2.9.4 Range Extension

In the case of extremely long station lines or high dc requirements for telephones the following additional measures can be taken:

- Extension increase for analog telephones by using the SLMAR board
- Extension increase for analog telephones with remote extension box
- Extension increase for digital system telephones with plug-in mains adapter

2.9.5 Interfaces

2.9.5.1 Interfaces to Subscriber Facilities

1. S_0 interface
   - According to FTZ guidelines 1TR210, 1TR211, 1TR212, and 1TR230, 1TR231 relating to layers 1 and 2.
   - 4-wire full duplex mode
   - Channel structure: 2 x 64 kbit/s for B-channel, 1 x 16 kbit/s for D-channel
   - Code used: AMI code
   - Layer 2 protocol: HDLC (LAP-D), ITU-Q.921
     a) S_0 interface as module in the system
     The interface is bus-capable and provides remote feeding.
     b) S_0 interface behind a PT (private termination) which is connected to the system via a U_{P0E} interface.
   - Terminal operation on the S_0 bus is possible with different function distribution (functional and stimulus) and different protocols (CorNet-T, , Q.931), with only 1 protocol per service, however. Direct connection to HiPath 4000 is via the , STMD (with or without feeding) and STHC (without feeding) module.
2. U_{P0E} interface

- According to ZVEI recommendations for layers 1 and 2
- 2-wire time division multiplexing
- Channel structure: 2 x 64 kbit/s for B-channel, 1 x 16 kbit/s for D-channel
- Code used: AMI code
- After conversion to S_0 using PNT, the interface is bus-capable
- Layer 2 protocol: HDLC (LAP-D),ITU- Q.921
- Connection to HiPath 4000 is via the SLMO or STHC modules

2.9.5.2 System Interfaces to Public and Private Networks

Connection of HiPath systems to each other in networks and their coupling to public exchanges is via the ISDN interfaces S_0 and S_2 using:

- CorNet NQ for HiPath 4000 networking
- Q-Sig protocol (ETSI or ISO standard) for heterogenous networks
- country-specific protocol (e.g. DSS1) required for service provider

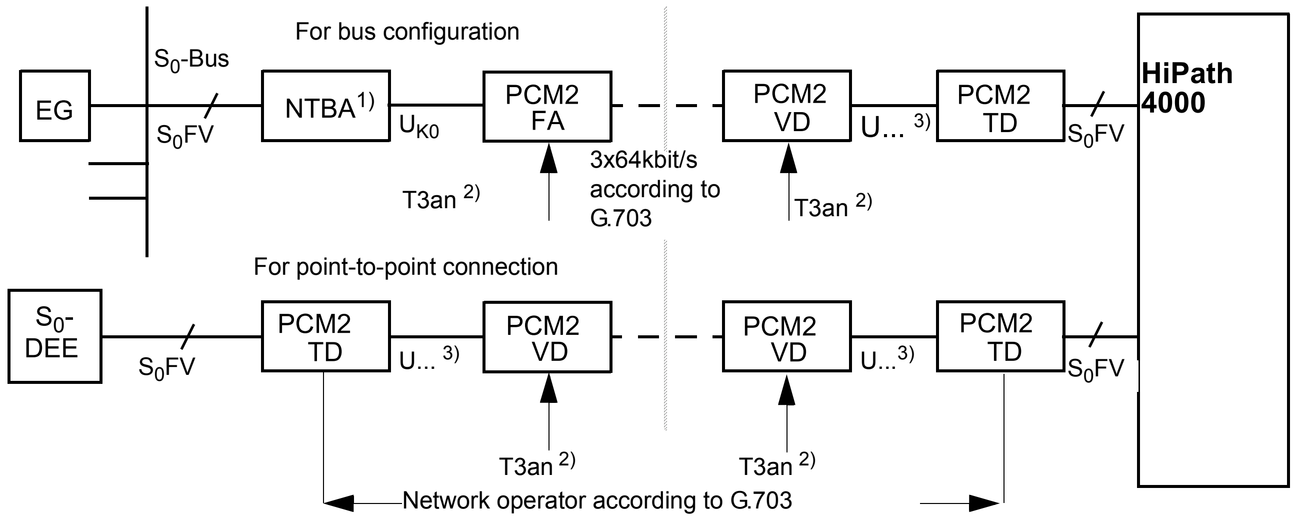
2.9.5.3 Off-Premises Extension

In HiPath 4000, the connection of off-premises extensions has been implemented for off-premises traffic.

If the connection between a terminal and the systemhas been made as a dedicated line over public property or across property lines, it is considered to be a remote station. The off-premises extension connected to the system via a dedicated line can set up dial connections to the partners behind the communication server.

Possible terminals:

- All terminals / services which can be operated on the S_0 bus.
- S_0 DTE in point-to-point operation.



FA: Foreign user access
FV: Nailed connection
NTBA: Network Termination for ISDN Basic Access
TD: Digital user terminal
VD: Digital switching terminal
1): The depicted type of operation for bus connections only functions if the "test loop" function is suppressed in the NTBA.
2): PCM2 VD has to be synchronised with the external T3an clock pulse:
- from the public network, if it supplies the HiPath 4000 system with the clock pulse,
- with the HiPath 4000 clock pulse, if the Hipath 4000 system is operated in private networks without connection to a public PABX.
For ranges greater than 250 m, it is not possible to synchronize the clock pulse between the PCM2 VD and PCM2 VD combination at a reasonable expense. Without clock synchronization frame slip has to be tolerated.
3): U interface, depending on the manufacturer (e.g. 2B1Q)

Figure 17 Off-Premises extension

2.9.5.4 Off Premises Station via T1- and SLMAR-interfaces (analog Connectivity)

Within this new function, the external interfaces to the Off Premises Station (OPS) could also be an T1 interface and a analog line interface .

All features and functions accessible to an analog station shall be accessible to an OPS station with the exception:

- Message Waiting Indication via additional lamp,
- T1 and the SLMA line interface
- The Feature Access for Off Premises Station (analog) will be activate/ deaktivate only by administration.

Message Waiting-Mailbox / Off Premises Station (OPS)

- Message Waiting Indication (MWI) via lamp shall not be supported on an OPS line (port), due to what is deemed to be hazardous voltage requirements.
- Message Waiting Indication shall be supported via inband indication (e.g., short announcement).

Disable flash for analog devices

It is possible to disable the flash capability from an analog device. It also includes a classmark to disable the detection of a hookswitch flash by an analog telephone.
When this classmark is assigned via a parameter used for such analog telephones, any attempt by a station user with that authorization to activate a hookswitch flash shall result in one of two states:

- An attempted hookswitch flash of short duration (less than disconnect time) is ignored
- An attempted hookswitch flash of longer duration (greater than disconnect time) results in an on-hook condition

2.9.5.5 High Performance UpO/E Line Card - SLMOP

The SLMOP (high performance) has the same functionality as the SLMO module.

It has the following characteristics-

- Use of a powerful microprocessor (state-of-the-art technology)
- Dedicated HDLC controller for each port.

Impacted devices

The complete digital system telephone family can be connected to the SLMOP. In contrast to the standard SLMO, routers which claim a D-channel exclusively for themselves can be connected as well (provided that no other terminal is simultaneously connected to the same port). These are devices with an S0 interface which are connected via a Terminal Adapter (TA-S0).

Interfaces

$U_{p0/E}$ Interface:

- Digital 2-wire interface
- Terminal with $U_{p0/E}$ or S_0 can be connected via TA-S_0
- -48-V remote feeding via $U_{p0/E.}$
- Max. cable length approx. 1000 m
- High performance capability (dedicated HDLC channels)

2.9.5.6 Mixed Line Card for So and U $_{pO/E}$ - STHC with 4 x S0, 16 x U $_{pO/E}$

The STHC is a multifunctional line card with a fixed port assignment. It includes both So and $U_{p0/E}$ interfaces, where the So can be used for both subscriber and trunk connections.

The following port allocation results from maximum 24 a/b wires that can be connected:

- $U_{p0/E}$: 16 ports via a 2-wire interface.
- S_0: 4 ports via a 4-wire interface.
- The assignment of the port type is fixed.
- The -40-V feeding to the subscriber via the S_0 interface is not supported.

Interfaces

The digital $U_{p0/E}$ and S_0 interfaces have the following characteristics: the given ranges are just guide numbers, cable specific values can be obtained from the ICCS.

$U_{p0/E}$ Interface:

- Digital 2-wire interface
- Terminal with $U_{p0/E}$ or S_0 can be connected via TA-S_0
- -48-V remote feeding via $U_{p0/E.}$
- Max. cable length approx. 1000 m
- High performance capability (dedicated HDLC channels)

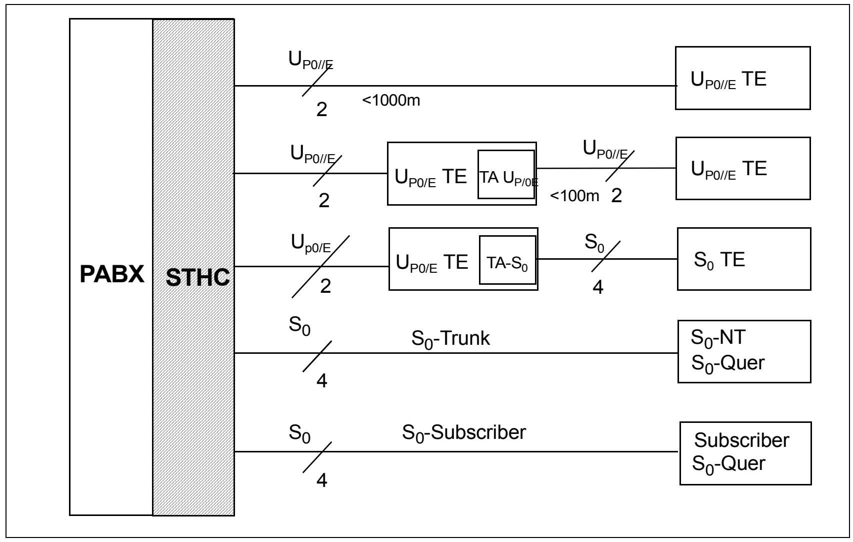
S_0 Interface:

- Digital 4-wire interface
- Trunk/subscriber (without feeding) mode
- Details in the trunk mode:
- Max. cable length in trunk mode approx. 1000 m
- Only point-to-point traffic is supported.

  Details in the subscriber mode:

  - No -40-V feeding
  - Maximum cable length in the subscriber mode
  - Extended bus: approx. 500 m
  - Short bus: approx. 150 m

- Both traffic types, point to point and point to multipoint are supported.



Connection of terminals to STHC

2.9.5.7 SLMAE Board

The SLMAE (Subscriber Line Module Analog Enhanced) board is an analog T/R interface (24 ports). They support the following analog devices:

- Announcement recorders
- Dial pulse or DTMF analog telephones
- Fax machines
- Modems
- Music-on-hold equipment
- Paging equipment

A code receiver is available for each analog interface (permits DTMF dialing at analog telephones). This guarantee that all analog terminals connected are fully accessible.

The board supports calling name identification presentation (CLIP).

This board generates its own ring voltages (71 Veff) and does not require an external ring voltage generator

Additional functions are:

- Overvoltage protection
- Ringing the line

– Supervising and signaling the line
– Codec function
– Hybrid function 2W to 4W and 4W to 2W
– Test (loopback) capability

2.9.5.8 SLMAR Board

The SLMAR board is characterized by the following features:

- 8 ports per board
- Fine protection on board
- HiPath 4000 (Sipac) connection method
- Metering pulses
- Expanded range for analog stations

Metering pulses

Analog card and coin phones can be connected to this board. 16 kHz and 12 kHz metering pulses (per port administration) and polarity reversal a/b are supported. The clock is either received from the superior CO or calculated internally.

The metering pulses of all stations in a system are stored in restart protected counters. There are 3800 counters per system.

The HiPath 4000 call charge recording can also handle analog metering pulses.

Range extension - electronic values

Maximum loop resistance (including station) 3000 Ohm with a minimum loop current of 18 mA.

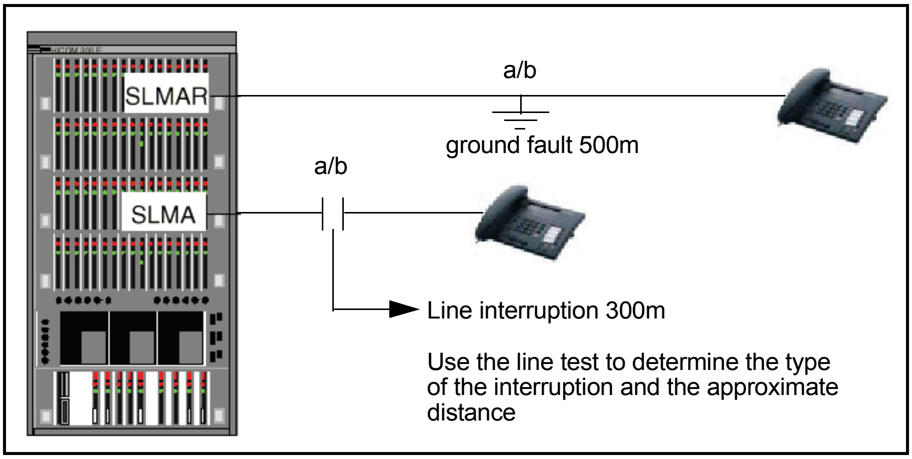
Feed characteristics: Impedance feed (range from 2 x 200 Ohm to 2 x 400 Ohm) and current limitation to 20 mA.

Two analog telephones per port maximum

The board does not support Message Waiting on analog telephones.

2.9.5.9 Simplified line check on analog subscriber lines

With HiPath 4000 this feature is used for the diagnosis of line faults (e.g. shorts or interruptions) on the a/b interface (on SLMA/SLMAR). In case of faults, this feature offers easy and quick differentiation for analog subscribers between the possible error sites ‘HiPath board‘ or ‘line network / station‘.



The error cause and site on analog subscriber lines are identified.

Requirements for and execution of the simplified line test:

- Only the SLMA and SLMAR - boards can be tested.
- The tests require an additional SIUX.
- The measurement frequency lies between 300 and 3400 Hz (DC measurement is not possible).
- The approximate line length should be known.
- Due to test tolerances and various unknown parameters the result is not always clear.
- All analog stations can be tested but only sequentially, one after the other.
- The feature can be activated both locally on the operating terminal and remote.
- The feature does not replace special measuring equipment for line tests.

Function and measuring principles

The signal generator injects a stimulation digital signal into the stations B channel. The SLMA converts this signal into an analog signal and sends it to the subscriber line. The a/b wires will receive a signal according to the line termination. This signal is received by the SLMA, converted into a digital signal and transmitted to a receiver where it is evaluated. The result is transmitted.

- The signal generator and receiver and is implemented on the SIU board.
- There are two different test jobs depending on the allocation status of the subscriber line:
  1. station idle: measuring in this state checks the presence of the clock capacitor.

   – Possible error causes: missing station, line interruption.

2. Station in timeout, i.e station is off-hook but without another party (off-hook without dialling, no on-hook after end-of-call): measuring in this state checks the AC voltage termination of the station.

   – Possible error cause: short-circuit on a/b-wires.

- Stations in talk state are not tested.

2.9.5.10 Analog trunk module (TMANI)

The analog trunk module TMANI offers a twin-wire interface to analog public trunks. The board includes the necessary circuitry to support loop closure or monitoring for call detection (loop or reversal polarity) and call charge detection for up to 8 ports.

The board supports:

– Outgoing traffic: direct

– Incoming traffic: by means of attendant console (MSI), direct inward dialling (DID)

– Bothway traffic

– Dial Signalling method: DP, DTMF and MFC-R2 (method configurable per circuit)

– CLIP for analog stations

- 12-KHz and 16-KHz call charge frequency band

2.9.6 Special Facilities

Special-purpose equipment is connected via the paging system (TMOM). The following devices and their service features are available:

1. Announcement unit
   - general announcement, for internal usage only
   - announcement with greeting text, if access is to be possible from outside (incoming trunk calls)

   Once the greeting text has finished, the system automatically switches to the general announcement.
2. Public address system
3. Entrance telephone

The following functions can be performed with different codes:

- Telephone/door conversation
- Open door

4. Paging system
   - Single paging system
     – with voice announcement with/without talk-back facility
     – with attendant callback (meet-me)
     – with display
   - Multiple paging system
     – with attendant callback (meet-me)
     – with display

The following applies to all special equipment operated with the paging system:

- override protection
- no suffix dialling in call and busy state except to the multiple paging steam.

2.9.6.1 Recorded Announcement Service

1. Recorded announcement service with random start
   - The caller hears the text beginning from a random point (continuous tape).
2. Recorded announcement service with pre-set start
   - The caller always hears the text from the start.
     Ring tone is heard until the start.
   - Recorded announcements with pre-set start are used primarily for greetings texts.
3. Connection of recorded announcement service to a trunk line
   - A recorded announcement service can be reached
     – by internal station users
     – by external parties
     – in consultation status.
   - The recorded announcement service is connected via a trunk group; this permits several callers to be connected simultaneously.

- As soon as the announcement code is dialled, the following callers are immediately connected via an idle announcement unit line:
  – internal users
  – external parties
- The caller can terminate the connection at any time by going on-hook.

4. Connecting an announcement unit via SLMA. Announcements can only be connected in asynchronous mode, i.e. announcements must be kept short, since they are played back in continuous mode.
   - An announcement unit can be connected to an SLMA, which allows an announcement to be made to each caller before switching through to an alternative station. This can be configured for all user types, but is primarily intended for attendant consoles or hunting groups.
   - The announcement unit is configured as an analog station, and is used for the AVOM feature (announcement before answering). After a few rings, callers hear a message recorded by the called party. As soon as the message has been played back, the announcement unit sets up a call transfer via flash or ground signal to a user in the same system (home node), or, in the case of networked systems, to a user in a foreign node, provided the nodes are connected via CorNet NQ or DPNSS1 links. The number of the station to which the call is switched is programmed in the announcement unit. The announcement unit transfers the call regardless of whether the station is idle or busy. By configuration (AMOs) a recall to the announcement device is inhibited.
5. Connection of announcement devices via TMOM.

   Announcements are made in synchronous mode, i.e. they are played back from the beginning in each case. This allows longer announcements to be played.
   - The zero pass signal from connected announcement devices is evaluated to allow through-connection at the same time as the announcement start when broadcasting. Incoming calls (currently only trunk calls) are queued until announcement start.
   - The areas of application for synchronised announcements are limited to:
     – Hold-the-line for calls to the AC if not answered
     – Announcement for call to multiple paging system
     – Announcement for call to a hunting group queue
     – Announcement for call to a busy station requesting suffix dialling
     – Announcement for call to a busy station with attendant intercept
     – Announcement for call to idle station requesting suffix dialling

  - Announcement for call to idle station with or without attendant intercept
  - Announcement for call to a busy station with automatic camp-on
  - Announcement requesting dialing digits for DID (only analog CO)
- Currently, up to 64 announcements (announcement units) can be configured per HiPath 4000 node, giving a possible total of 64 announcement types. Up to 50 callers can be switched to one announcement device. More than one device can be configured with the same announcement type to accommodate more than 50 callers. In extreme cases all 64 announcement devices can be allocated to only one announcement type.

2.9.6.2 Voice Paging Access (Public Address System)

Interface for the connection of a public address system with amplifier with one or more loudspeakers.

- An announcement over the public address system can be activated using the standard procedure.
  - It can also be seized in consultation status.
  - The announcement over the public address system can be ended by releasing the connection to the calling station.
- A maximum of 100% of users can be authorised to make announcements over the public address system.
- Users are given the appropriate class-of-service with the aid of administration and maintenance.
- The voice paging system can be seized by
  - internal station users,
  - the attendant console if the public address system is not connected via TMOM

2.9.6.3 Entrance Telephone

Interface for the connection of an entrance telephone with door opener function.

- The entrance telephone can be answered from every station in the system by dialling the appropriate answer code.
  - Answering is also possible in the call state (consultation with the entrance telephone).

  - The answering call is terminated by releasing the call or withdrawing the consultation to the calling station.
  - If two or more station users are answering, the station user which first dialled the code is given the connection; the other station users receive a negative acknowledgment.
- During the answer call, the door can be opened by dialling the opener code.
  - It is possible to dial the door opener procedure following answer during the call state.
  - The call state is also retained after the door opening procedure until the station user releases the connection.
- All station users are authorised to answer the entrance telephone and to open the door.
- The entrance telephone can be seized by internal station users,
- The door can be opened using the door opener procedure by internal station users.

2.9.6.4 Paging System Access (Code Calling System)

Station users who temporarily cannot be reached by telephone can be paged by means of a paging system if a paging option is configured for the dialled call number.

1. Universal interface for the connection of paging systems.
2. Conversion of dial pulsing, push button dialling, digital selection using pulse dialling or DTMF to the interface.
3. Simple paging system
   - Connection of a visual paging system with attendant callback
   - Connection of a wireless paging system
     - with answering callback (meet-me)
       User/attendant being paged receives alerting tone. The requested call connection is established by dialling the paging answering code at any station.
     - with voice announcement
       without the possibility of talking back
       with the possibility of talking back
     - with display facilities
       User/attendant being paged receives a display of the paging user's call number.

- Only one paging job can be carried out at a time.
- If conversion of the station call number into a paging number is necessary, this takes place in an adapter connected before the paging system frame.
- The transmission method (pulse dialling or DTMF) for the data is in accordance with the paging system frame or the pre-connected adapter.
- The dialling aid keys of the analog and digital voice terminals can be used locally and on a network-wide basis to activate this feature. The following paging variants are possible:
  - Callup of a complete digit series
  - Callup of a partial digit series with subsequent manual dialling of the remaining digits
  - After initial manual dialling, callup of the remaining digits using a dialling aid key.
- Simple paging can also be set up as a hotline or off-hook recall destination.

4. Multiple paging system

   - Connecting a wireless paging system
     - With "meet me". The requested connection is made from any HiPath 4000 station by dialling the answering code followed by entering the user's own number.
       The paging station can page within the network with either a lifted or cradled handset. Paging functions network-wide only when the handset is lifted.

       The call number of the code-calling (paging) party (A-block) is transmitted to the called (paged) party. The complete search string, which consists of the A, B and C-blocks, the block separators and the end-of-block character must not exceed 22 characters, since otherwise the A-block information will be truncated.

       Exchange calls which are routed to the CCM (meet-me) by means of variable call forwarding or forward-on-no-answer are indicated differently to internal calls, i.e. by means of an "urgent" ringing as opposed to the standard ringing.

       The standard signal transmitted to callers during a "meet-me" code call is an idle or ringing tone for exchange or tie-line callers, and either ringing, knocking or music-on-hold for internal callers.
       Also a voice announcement can be made in place of the standard signalling tones.

  - With display
    (Number of the paging station).

- As described above, a search job for a multiple code-calling system basically consists of the A-block, B-block and C-block, which are transmitted in this sequence.
  The blocks have the following contents:
  - A: Call number of searching party
  - B: Call number of called party
  - C: code digits.

  In compatibility with other code-calling/paging systems, an alternative "B-C-A" block transmission sequence can be set via administration and maintenance.

- A maximum of 15 paging orders can be stored simultaneously.
- The paging jobs are carried out consecutively by the paging system at the correct times.
- Two or more paging jobs can be stored at the same time for one paged station user.
- With the administration and maintenance facilities it is possible to stipulate whether paging is to take place with or without call number/ paging number translation.
- If paging takes place with call number/paging number translation, a paging request is only issued if the wanted party also has a paging number.
- A maximum of 1,000 users can have paging numbers.
- Paging using
  - prefix dialling,
  - suffix dialling to the paging system in the case of idle/busy for originating and consultation calls (not for attendant console).
    The code calling destination number does not need to be a HiPath 4000 station number, but must not be any longer.
  - Consultation.
- Call forwarding - all calls and call forwarding - no answer to the paging system for the "meet-me" service in internal, external and attendant traffic in originating and consultation calls.
  Call forwarding - all calls/call forwarding - no answer is **not** possible from the night answering station in the night switching condition.
  The code calling destination number does not need to be a HiPath 4000 station number, but the length of the number must be compatible.

- Cancellation procedure for the paging request (only attendant console).
- Optional continuation of paging when the paging party goes on hook (internal only)
- DID to the paging system.
  An external party can access the paging system using prefix dialling.
  If required, this function can be combined with attendant intercept.

5. All subscribers are authorised to use the paging system.
6. All attendants are authorised to use the paging system.
7. Switching exchange connections
   - Calls can be switched to the paging/code calling system by users or by the attendant on a network-wide basis.
   - The definitions for switching outgoing exchange connections to a multiple code-calling system are the same as those for switching outgoing exchange connections to users in calling state by the attendant in basic traffic.
     Until the required party answers, the call charges are assigned to the attendant position. Once the required user has answered the code call/ paging call (at any station), the connection charges are transferred to the user.
     Restrictions:
     – Only dialled (established) exchange connections in talking state can be switched
     – Existing FWD/FNANS destinations programmed by the user are ignored
     – Lines cannot be transferred before party answers
     – Lines cannot be transferred if connection established via attendant intercept, recall or cut-through dialling
     – Lines cannot be transferred by centralised attendant.
8. Group calls are possible (multiple paging) for single-call and multiple call code-calling systems (paging systems) in "Meet-me" or "display" operation, as well as for single-call CC systems in "voice announcement" mode. The group code-calling number is assigned in the CC (PSE) system. The users of the group are called simultaneously via this number.
9. A paging code is assigned to the paging key. This code is determined by the digit interpretation group to which the station user belongs.
10. Once the paging key has been pressed or a paging code has been dialled, the paging key LED is lit until the paged station user answers or the call of the person paging is transferred/switched or until the paging process is terminated.

11. Station users who do not have their own telephone station can be paged with the paging system.
    - Users without their own station receive an individual paging code. In this case, paging can take place with prefix dialling and in consultation.
12. Answering by the paged station user
    - The paged station user can answer from any desired station using prefix code calling.
    - In multiple paging systems, the paged station user must dial his own call number (or his own paging code if he has no call number) in order to identify himself.
13. In the case of multiple paging, the paging process is terminated after a defined, pre-programmable time if the paged station user has not answered within this time. The time is dependent on several parameters that can be controlled via AMOs
    - If the paging station user is waiting with his handset off hook, he receives busy tone; the paging key LED goes off; there is a special paging system timeout.
    - If the paging station user is waiting for an attendant callback: the paging key LED goes off; there is a special paging system time out.
    - The duration of the timeout for termination of paging is defined by means of the administration and maintenance system.
14. Paging in consultation in the 'display' operating mode
    Users connected to a multiple paging system can also be paged in consultation if the following conditions are met:
    - display' operating mode,
    - paging/paged user in one node.

    This feature can also be implemented on a network-wide basis if the
    - paged user and multiple paging system are located in one system (A) and the
    - paged user answers from system (B) and
    - System (A) and (B) are networked via CorNet-NQ.

    The answering procedure is only possible from an originating call. The feature does not apply to the attendant console.

2.10 HiPath Cordless

HiPath cordless is an integrated radio switch solution for HiPath 4000.

2.10.1 SLC24 Module with Integrated Radio Switching

The SLC24 module handles the radio switching functions and the subscriber database. It also allows DECT base stations to be connected via a $U_{P0/E}$ interface. The CorNet-TR protocol is used, ensuring that HiPath features are also available for cordless subscribers. Several SLC modules can be configured in a HiPath 4000 system. Roaming (base station reregistration while idle) and handover without interruption (base station reregistration while conducting a call) are possible between base stations.

2.10.2 Base Station (DECT)

The base stations form the radio cells and thus support communications with the cordless terminals, while also connecting the "air" (wireless) interface with the line interface of the SLC. The base stations can be connected to the switch via 1-3 $U_{P0/E}$ interfaces. When fully equipped, the 12 DECT channels (time slots of the air interface) of the wireless interface which are available per base station can be switched through to the system completely.

The maximum range of the $U_{P0/E}$ interface is 1 to 2 km, depending on the type of cable used. The HiPath 4000 system feeds the base stations.

2.10.3 Features

1. General features

   The integration of the radio switch allows cordless subscribers to enjoy all relevant HiPath 4000 features by using an easy and comfortable dialog-oriented user interface.

   Major features of HiPath cordless E:

   - Digital wireless technology based on DECT incl. GAP (Generic Access Profile: air interface using the DECT standard)
   - Standard and comfort handsets from the Gigaset product family
   - Base stations utilizing all 12 DECT channels (fast hopping technology), which can be switched through to the SLC line interface without blocking
   - Outdoor housing for base stations
   - SLC module with integrated radio switching
   - Roaming and handover between base stations which are connected with the system via different modules. Network-wide roaming over up to 32 HiPath 4000 systems.
   - Administration and maintenance via HiPath 4000 Manager

The company-specific expanded GAP air interface supports additional user-friendly features such as control of the display.

DECT technology allows radio cells to be realized with high subscriber density.

The range of the base station is up to 50 meters in buildings with concrete walls and up to 300 meters outdoors, despite the low transmission strength of 10mW. The multicell technique is used to achieve full coverage by overlapping the individual DECT cells. This allows handover and roaming without interrupting calls, as mentioned above.

2. Features realized analogous to the digital system telephone user interface
   - Connection setup and release (basic call)
   - Display of call number and name of caller
   - Display of dialled number (local echo of digits entered)
   - Display of call charges on mobile unit
   - Consultation call (second call)
   - Transfer before and on answer
   - Conference
   - Callback
   - Terminate second call and return to first
   - Cancel dialling and return to ready-to-dial mode
   - Cancel outgoing call and return to ready-to-dial mode
   - Alternating (toggling)
   - MWI
3. Features available via code procedure, independent of analogy with the digital system telephone user interface:
   - Individual speed dialing
   - Central speed dialing
   - Hotline
   - Switchover to DTMF dialing (via code)
   - PIN
4. Features which are in part analogous to the digital system telephone user interface:
   - Feature: Executive / Secretary system (CHESE)

A mobile unit can not be a member of an executive/secretary system. Any subscriber, i.e. including an executive or secretary, can activate call forwarding to a mobile unit.

- Mailbox and start (output) keys

  An announcement "Message waiting" indicates that a voice mailmessage is waiting, as with analog terminals and with MWI. The user can play back messages using the number keys of the keypad, as with an analog terminal.

- Fixed redial key

  Redial is controlled from the local mobile unit.

- Key for activating fixed call forwarding

  The call forwarding feature can be used by mobile units. The user can forward calls to any authorized HiPath 4000 destination (Anate, Digite, etc.). Fixed call forwarding is programmed and activated by entering a code on the mobile unit.

- Call pickup key

  A call for a member of a call pickup group is not signalled to a mobile unit. However, call pickup can be activated on a mobile unit by entering a code.

- Hunting group

  Mobile subscribers can be members of a hunting group. The hunting group can be cyclical or linear. A function key is required to activate/deactivate participation in a hunting group. Mobile units do not have such a function key but the function can be activated/deactivated using AMO’s at the service terminal.

- Knocking / busy override by key

  The knocking or busy override key is not supported on mobile units, but these functions can be activated by entering a code.

5. The following features are not available:

   - Disconnect (release) key

     The disconnect key is not available on the mobile unit.

   - Messenger call key

     The messenger call key is not available on the mobile unit.

   - Repertory, DSS and timed reminder keys

     These keys are not available on the mobile unit .

   - Park and continue parked call

The fixed keys required for this feature are not available on the mobile unit.

- Loudspeaker announcement
- Secretarial intercept, deputy circuit

  These features are not supported for the mobile unit.
- Voice calling

6. Features which are controlled by the SLC24 module:
   - Activate/deactivate transfer of ringing
   - Suppress display
   - Call waiting - terminating
   - Class-of-service changeover
   - Speed dialling individual/central
   - Class-of-service follow-me (PIN)
   - Query call numbers of unsuccessful call attempts
   - Activate diagnostic mode via special access procedure
   - Mute (with soft key)
   - extended missed calls
   - HiPath DTB support
   - Direct respond
   - Calling tone signaling internal/external
   - After switching on the terminal, entering the functions area and the applying as Cordless attendant, the time is automatically set on the Gigaset by the system.

2.10.4 Frontreference

In the case of HiPath 4000, the frontreference can be used in every IPDA. This is important in connection with Cordless and ISS (i.e. seamless handover) because the time information for ISS can also only be used in the case of precise clocking from the GPS.

All base stations of an Cordless System (CMI) have to be synchronized according to the DECT requirements to enable the seamless handover independent whether they are connected to an IP access point (AP 3300 IP, AP 3700 IP) or to a host system shelf (AP 3300, AP 3700). Therefore the IP access points need to get synchronization signals via the LAN infrastructure.

This requirement is realized by implementing the Precision Time Protocol (PTP, IEEE1588-2002) in the IP gateway HG3575 V4 (NCUI4). This implementation allows clock synchronization over the IP network. The clock master is one of the APs (HG3575 - NCUI4), if the host system consists of the communication server only. Now the IP access points AP3x00 IP can run with control board NCUI4 to realize the CMI handover between CMI base stations connected at two IP access points.

2.11 Additional Technical Features

2.11.1 Baby Phone

The HiPath 4000 allows the use of the Baby Phone function, e.g. in hotels when a guest wants to leave the room without the baby. The guest can control the room where the baby is from a different station within the same hotel PABX by dialing a special code.

The feature is activated when a special code (and, if necessary, an optional identification against misuse) is dialed on the station of the hotel room. The handset remains off-hook beside the station. It is now possible to call this station from a different telephone in the hotel. The caller hears the busy tone. By dialing a special code the connection is made to the hotel room. The guest can now hear the baby.

2.11.2 Sequential Dial Tone Supervision

For external dialling in the course of which dial tones can be expected when specific setup stages have been reached (e.g. after the trunk-access code, in some countries before dialling the country code) can be used to set up waiting for these tones.

Until the trunk circuit has been connected (using the example of an outgoing trunk call with dialling of the trunk-access code), within a max 18-position digit string it is possible to administer what are called "dial tone markers" at any 6 positions. A dial tone marker signifies that within the transmitted dial string a dial section has been reached after which the caller must wait for a tone.

For the dialling of externally dialled digits, a "dial tone marker" can be administered after each digit within the max permissible 22.

If no tone is received in situations in which a dial tone is expected in keeping with the administered dial tone markers, the connection is released after a definable waiting period and the user requested to repeat his inputs.

There is no tone evaluation. It is assumed that the received tone is the expected dial tone.

2.11.3 Transmission Level Monitoring

For use in special networks, the level sent by the distant system is echoed back by the HiPath 4000 system on 2 separate signalling lines. The transmission level monitoring takes place on E&M tie-lines (TMEW2), for continuous signalling and pulse signalling.

Transmission level monitoring is provided in both directions. There is no active signalling in the outgoing direction.

A drop or failure of the signal level received from the remote system of more than 7 sec. duration results in remote blocking of the relevant incoming tie-trunk. An error message is printed out at the service terminal. The remote blocking is deactivated as soon as the correct signal level is detected again over at least 2 seconds.

2.11.4 Music on Hold via "External" Announcement Unit

Internal or external callers (station users and attendants) on hold normally receive an audible notification signal from the SIU. The parameters of the notification signal can be set via the ZAND AMO for the entire system.

If alternate signalling is desired, an "external" announcement unit can be connected to any SLMA . This enables system owners to use individual notification tones or announcements.

The following requirements exist for connecting the external announcement unit:

- For each type of signalling (announcements or tones), a separate SLMA connection must be configured per LTG,
- The SLMA connection can be configured as required,
- 8 ports are available per announcement unit, allowing 8 LTGs to be supplied with individual tones
- Max 5 tone types (= timeslots) can be replaced by individual announcements or tones. The assignment must be identical in all LTGs.

2.11.5 Staggered Dialling and Suffix Dialling in Analog Telephone Networks

Outgoing connection setup via a digital tie-line with access to a conventional, analog telephone network can be staggered. The caller can continue dialling

- after listening to an announcement in the analog network (staggered dialling). This can be repeated several times with different announcements in the course of a connection setup.
- after the HiPath system has received the answering criterion (suffix dialling).

This feature is intended for

- digital and analog station users
- stations with pulse dialling or DTMF dialling
- HiPath system users with direct access (gateway) to a conventional, analog telephone network via a digital tie-line (with CAS),
- for HiPath users with indirect access to a conventional, analog telephone network via one or more HiPath 4000 systems (transit nodes with S2-networking).

Staggered dialling and suffix dialling are features which are primarily used in the analog networks of public transport services in Germany.

Staggered dialling

The user selects the route to the call destination directly, by dialling the access codes of the transit switches and the destination switch according to the network plan. For checking purposes, each switch plays back a local announcement, which informs the user that the call setup is proceeding as desired.

In order to play back the announcement to the user, the following paths are switched:

- users with DTMF telephones: in order to avoid interference by the DTMF receiver, only the ear channel is switched.
- users with other telephones: both the ear and the mouth channel are switched.

In addition, around 15 seconds are permitted between dialling in order to listen to the announcements. Users with DTMF telephones remain connected to the DTMF receiver for 15 seconds or until the answering criterion is received.

Suffix dialling

Normally, the answering criterion indicates that the called party's line has been reached and that the called party has answered, so that no more dialling information is necessary. This is also the latest moment in the HiPath 4000HiPath

4000 system at which, in addition to the ear channel (for DTMF telephones), the mouth channel is switched through from the user to the seized outgoing line (through-connection). However, in the public transport services' networks, there are exceptions, such as:

– connection switching for local battery leased lines

– connections to RF transceivers (Funk - UELE).

Connections of these types require the seizure of special line circuits, from which new dialling information must be transmitted in order to reach the line circuit of a remote user. Since the answering criterion cannot be transmitted when the actual called party answers due to the special line circuits in between, the answering criterion is simulated immediately after seizing the line, i.e. before the user is reached.

Users with DTMF telephones remain connected with the DTMF receiver for up to 15 seconds after they have dialled the last digit in the sequence, regardless of whether the connection is already established or not.

Once the 15-second-timer has run out, the dialling phase is regarded as completed (simulated end-of-dial), and the system will not accept any further dialling information.

Terminals from which staggered dialling or suffix dialling is possible

– Analog tekephones with DTMF dialling

– Analog telephones with pulse-dialling

– Digital system telephone

– IP- telephone

- Staggered dialling and suffix dialling is not possible from the attendant console.

- In transit systems, the connection path is switched through immediately after the outgoing route is determined and seized. The end-of-dial signal is not simulated at the end of the 15-second timeout; the lines in the transit node remain open for dialling information.

- In the gateway, the transition to the analog network from the incoming DIUT2 to the outgoing DIUN2 is achieved by protocol conversion CorNet NQ ←→ CAS (single bit / multiple bit protocol). Here, too, the connection is switched through immediately the outgoing route is determined and seized. The end-of-dial signal is not simulated at the end of the 15-second timeout; the lines in the transit node remain open for dialling information.

- From the moment the answering criterion is received, up until the end-of-dial timeout, a three-way connection exists between the mouth channel of the DTMF telephone, the mouth channel of the outgoing line and the DTMF receiver.

2.11.6 TFZ Signalling

TFZ stands for Trägerfrequenz mit Zwangslauf, and is a carrier frequency signalling system with forced frequency passage, which is widely used in the private analog networks of public transport authorities in Germany.

Such networks have a digital access point via which a HiPath 4000 system can be connected. PCM30 highways are used for channel-associated signalling. The use of a primary rate access multiplexer (e.g. Siemens PCM30-G with E&M interfaces) allows signal conversion to analog networks via carrier frequency signalling or analog interfaces.

1. TFZ signalling is used for lines which require a high standard of failure monitoring. These lines are linked via dual passage carrier frequency systems, i.e. with high-passage and low-passage signalling.
2. The following TFZ features ensure that the safety requirements of the analog lines are met:
   - Low-passage monitoring in idle state through transmission and detection of a permanent low-passage signal. Failure of this signal indicates a line failure or transmitter failure on one of the speech paths of the line.
   - Forced frequency passage for
     - line seizure
     - connection release (forward and backward), and
     - blocking and unblocking
       using both the high-passage frequency and the low-passage frequency. This ensures that the response of the line terminating equipment can also be monitored during transition from or to idle state.
   - Line monitoring after seizure (i.e. in busy state) is left to the user (audible tones while call is ringing).
3. In order to distinguish between the high-passage frequency and the low-passage frequency on the analog side, each TFZ interface has two inward signalling wires and two outward signalling wires. This applies to all the interfaces, i.e. those of the telephone systems, the carrier frequency systems and also those of the primary multiplexer (analog side). These latter also convert the signalling information and speech pulses from analog signals to digital signals and vice-versa.
4. The signal conversion in the primary multiplexer is carried out as follows:
   - high-passage frequency ↔ bit a of timeslot 16
   - low-passage frequency ↔ bit b of timeslot 16

   The time-dependencies of the signals are maintained.

Depending on the carrier frequency transmitter, the low-passage signal (a=1/b=0) can only be accepted as steady after 60 ms. All other signals, such as

- transition from zero-passage to high-passage,
- transition from low-passage to high-passage,
- transition from high-passage to zero-passage and
- transition from high-passage to low-passage

can be accepted as steady after 16 ms at the most.

5. If the analog path (or part of the analog path) is linked via dual-passage carrier frequency transmitters, then the telephone systems must assume that high-passage signals and the low-passage signals may be transmitted simultaneously. This is taken into account when defining the analog signalling criteria, i.e. the receiving side will interpret a high-passage signal and a high/low-passage signal as the same. The HiPath 4000 system will evaluate bits a and b similarly.
6. Signalling is carried out symmetrically
7. The analog line uses pulse dialling criteria exclusively.
8. Each single line can be configured for
   - bothway traffic
   - for outgoing traffic only, or
   - for incoming traffic only.
9. Each individual line can be blocked against seizure by the remote switching network (i.e. lines configured for bothway or incoming traffic in the local switching network and for bothway or outgoing traffic in the remote switching network). The seizure block can be initialised when the line is busy, but does not take effect until the ongoing connection is terminated (soft blocking).
10. Each channel of a DIUN2 can be configured separately for TFZ signalling.
11. Other channels of a DIUN2, which are not configured for TFZ signalling, can be configured for standard carrier frequency signalling, E&M signalling or EB5/NAL signalling.
12. TFZ signalling is not used on exchange trunks (for exchange traffic).
13. The features
    - staggered dialling or suffix dialling
    - backward release if line busy
    - switch dialling tone if line busy

    can be assigned per trunk group to channels with TFZ signalling.

14. If errors are detected, such as
    - no release acknowledgement (missing low-passage signal),
    - no low-passage signal after placing in service,
    - no low-passage signal detection in idle state or
    - circuit permanently seized,

    the system transmits test signals of 1s length every 10 - 30 s, until the low-passage signal is detected again.
15. Busy channels (from initial seizure to connection release) do not carry out a low-passage signal monitoring.
16. Dialling information feedback on outgoing channels is ignored.
17. Incoming dialling information on busy channels is ignored.
18. Line signalling criteria
    - Idle state (channel is ready)
    - Initial seizure
    - Busy
    - Pulse dialling (60/40 ms)
    - Answering criterion (80 ms pulse)
    - Connection release, unblocking
    - Blocking criterion (60 ms pulse)
    - Test signal (1000 ms pulse)
    - Not ready (prior to placing in service, and in the case of malfunctions).
19. Low-passage signal monitoring

    This allows line terminating devices to monitor analog lines in idle state (as long as they are linked via dual-passage carrier frequency transmitters).

    A low-passage signal is permanently transmitted over each idle carrier frequency channel (i.e. each analog line). The line terminating equipment (transmitter/receiver) at both ends of the analog line monitor the low-passage signal on the idle line, and block the line for outgoing seizure if the signal fails. Line alarms are then signalled, and test signals are transmitted until the low-passage frequency is detected again.

    In the case of digital interfaces from a HiPath 4000 to the analog paths, i.e. primary rate access multiplexers, monitoring is also carried out via the digital interfaces to the system. In addition, each single digital channel is monitored via bit c of the signalling channel (c=1 is set on error).

2.11.7 Exchange Call Recovery if Station "Not Ready"

Devices such as station line circuits, trunk circuits etc. were immediately blocked by the dependability system and marked as "not ready", if errors were detected by the error analysis system.

Error types can be:

- station line circuit or trunk circuit errors
- terminal errors (e.g. message flooding due to hardware error or due to user "playing" with key functions etc.).
- physical line interruptions between terminal and station line circuit.

Within HiPath 4000, the above errors are treated as follows:

- exchange call are re-rung at the attendant console, if a terminal or line circuit is taken out of service during the call
- internal and non-voice calls are terminated via backward release.

2.11.8 Synchronising to Active Layer 1

Within HiPath 4000, the board loadware differentiates between layers 1, 2 and 3 of the ISDN OSI protocol system, and the reference clock synchronisation is carried out as per ITU requirements. Since the Layer 1 signalling is now detected separately by the loadware, a failure of the layer 2 protocol will not automatically result in the loss of the reference clock.

- Failure of the layer 2 protocol does not result in the loss of the reference clock.
- Layer 1 errors, which do not affect the reference clock do not result in a loss of synchronisation.
- Layer 1 setup is detected by the loadware and synchronisation is started immediately.

2.11.9 Test Status Output for Hardware Components

With the aid of the routine test order of the dependability system, defective hardware components of the HiPath 4000 system can be detected and blocked by the error analysis system of the dependability software (if necessary, together with associated components). The test results of the RTO are only output if the error analysis system reacts.

For some errors (e.g. sporadic errors), a detailed analysis of the error event is only possible with the TSU AMO. The TSU AMO calls specific routines of the RTO in order to test the specified components, and outputs both positive and negative test results. Users of this feature should note that a negative test result is also output if a component is in operation (and therefore cannot be tested).

The TSU AMO also includes the test status of the component in negative result outputs.

2.11.10 Mute Function (digital system and IP-telephones)

The mute switch has been implemented as a key function. The key function is configured per AMO in the key layout standards of the terminals. Users cannot program this function directly.

By pressing the Mute key during a call in talking state, the user switches the handset and the handsfree talking microphone off. The LED on the key lights up. For switching the handset and the microphone back on, the user simply presses the Mute key once more. The mute function is not signalled to the remote party, i.e. the user will have to advise him or her of the function before activating it.

2.11.11 T-Reference Point (Digital Attenuation Value Settings)

HiPath 4000 supports the "T-reference point" feature. This feature ensures that all the digital voice terminals released for use in the HiPath 4000 system are operated at the same, optimum transmission levels, with a minimum of transmission echo.

1. The T-reference point is a defined interface point between a system and the public ISDN network, at which pre-defined voice signal levels are achieved on all lines. Its position in a system corresponds to that of the main distribution frame of the system.
2. In order to achieve a constant voice signal transmission quality, regardless of the connection type and the network size, the T-reference point conditions are also applied to analog interfaces.
   - This affects all connection types, in which the following trunk/circuit types participate:
     – Exchange lines (CO trunks), both analog and digital, to public ISDN network local exchanges,
     – Tie-lines, both analog and digital, for inter-PABX traffic

– Station lines (subscriber lines), both analog and digital, i.e. circuits to which the user terminals are connected

2.11.11.1 Localisation of the T-Reference Point

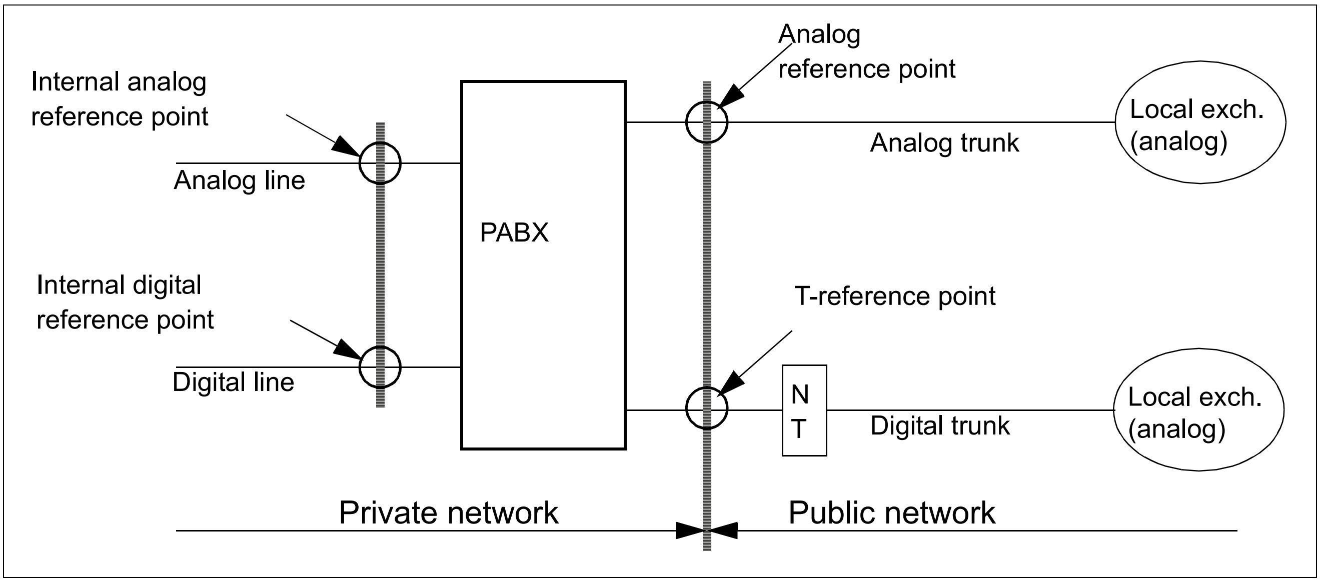


Figure 18 Local assignment of the T-reference point within the PABX (measuring points)

For digital exchange connections, the T-reference point is on the user side of the network termination, and corresponds to the position of the MDF. For analog connections, the reference point is also considered to be the MDF, i.e. the attenuation values of the exchange line are not taken into account.

In order to meet the transmission level requirements of the T-reference point matrix (loadable values) in all cases, i.e. even within private networks with different types of public network access, a so-called "internal reference point" is established for both digital and analog lines.

Since all the HiPath 4000 compact systems are adjusted to meet the level requirements of their respective T-reference points, and since these are identically defined in the local T-reference point matrix, groups of compact systems can be networked without any special signalling adjustments.

2.11.11.2 Use of Attenuators and Line Boosters for T-Reference Point Levels

In order to achieve the precise, required voice signal transmission levels at the T-reference point, a number of attenuators and boosters are available, which can be looped into the LTUCE voice transmission channels (-4 to +12 dB). These are switched across the appropriate ports of the LTUCE in response to the T-reference point value determined by the call processing system as required, in the receive direction.

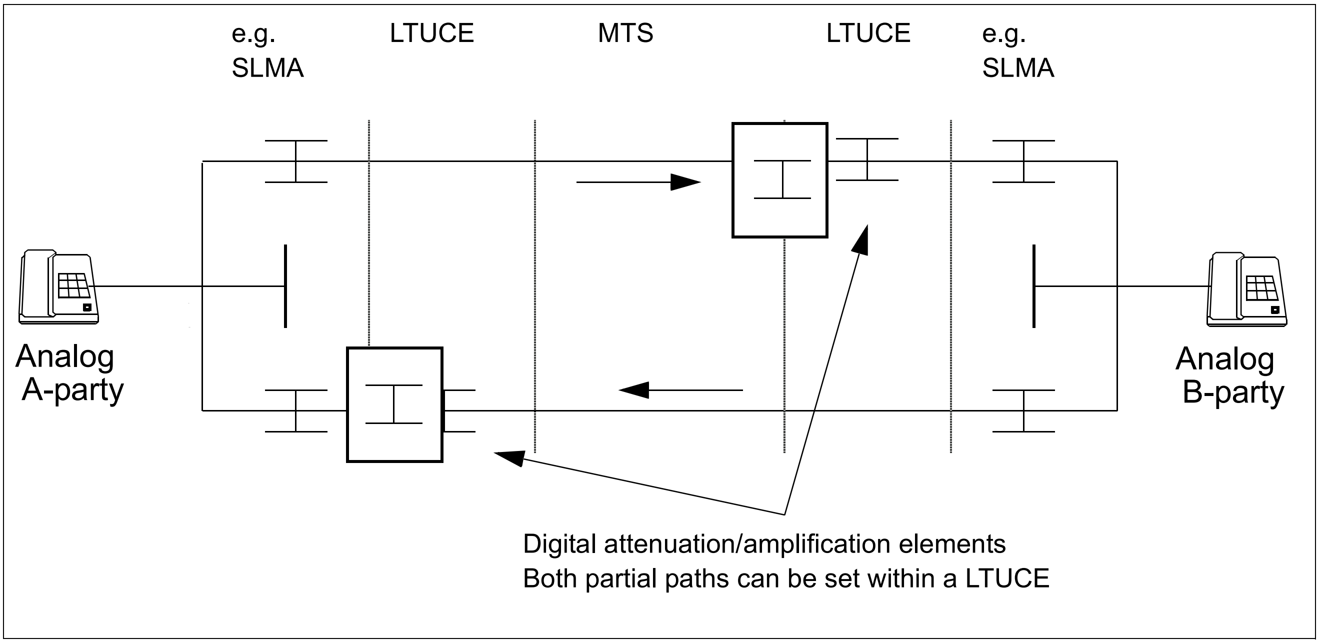


Figure 19 Principle arrangement of digital attenuation/amplification elements within a HiPath 4000 system (setting points)

This half-path solution allows different level values to be achieved for each transmission direction.

The T-reference point value for the connection being set up is read from the T-reference matrix, which is loadable. The T-reference matrix is country-specific; the following diagram shows an example of the T-reference matrix for Germany. The appropriate value is then transmitted back to the LTUCE.

From \ To	undefined device	Anate with SLMA	Level A Digite *)	Level B Digite *)	Analog trunk	Digital trunk	2-wire tie-line circuit,	2-wire tie-line circuit,	Tie-line circuit,	Tie-line circuit,	reserve
undefined device	0	0	0	0	0	0	0	0	0	0	0
Anate with SLMA	0	7	8	3	0	3	2	0	3	3	0
Level A Digite *) (high-volume)	0	8	10	4	1	4	2	1	4	4	0
Level B Digite *) (low-volume)	0	4	5	0	- 3	0	0	- 3	0	0	0
Analog trunk	0	0	1	- 4	0	0	0	0	- 4	- 4	0
Digital trunk	0	4	5	0	0	0	4	0	0	0	0
2-wire tie-line circuit, analog internal calls	0	2	2	0	0	3	0	0	3	3	0
2-wire tie-line circuit, analog trunk calls	0	0	1	- 4	0	0	0	0	- 4	- 4	0
Tie-line circuit, 4-wire analog	0	4	5	0	- 3	0	4	- 3	0	0	0
Tie-line circuit, digital	0	4	5	0	- 3	0	4	- 3	0	0	0
reserve	0	0	0	0	0	0	0	0	0	0	0

*) The values entered in the marked coordinates are transmission values for "loud" or "quiet" telephones. In a PABX system, each known digital terminal type is assigned its own coordinates.

The call processing system checks the appropriate bit for the T-reference point feature in the central system feature bit string. If this bit is set, i.e. the feature is enabled, the call processing system interrogates the "attenuation classes" for voice terminals which use the bearer services "speech and 3.1 kHz audio", and derives the appropriate transmission level values per type of connection. The T-reference point level of a connection is reset each time an information element is received to indicate that a status change has taken place (e.g. change of service, consultation call).

Once the attenuation classes have been determined by the call processing system, the appropriate levels are assigned in each case, according to the T-reference point matrix as shown above. If the T-reference point matrix does not contain any specific values for an attenuation class, the level values is assumed as 0dB.

The attenuation values for the voice signal levels are country-specific. All the country-specific T-reference point matrices are stored in a TREF table in the PABX hard disk. The appropriate values are loaded into the database via the TREF AMO, specifying the required language/country-identifier.

The T-reference feature bit is also activated or deactivated with the TREF AMO.

The coding standard (A-law, μ-law) for the LTUCE is still administered with the ZAND AMO. This setting is not changed during operation.

2.11.12 MFC-R2 Signalling

The R2 multifrequency signalling code is a ITU standard signalling method for international communication. This method is described in the ITU Blue Book standards Q.400 through Q.490 (Volume VI, Fascicle 4). The ITU recommendations leave some options open for country-specific interpretation.

The signalling method differentiates between analog and digital line signalling and register signalling. The register signals are defined in Q.440 through Q.490. The register signals provide dialling information (user information) for setting up connections.

The R2 line signals are signals which are transmitted between line repeaters/ terminations (link-by-link). These signals provide the line monitor criteria (idle state, seizure, seizure acknowledgement, answering, charge pulses, forward release, backward release, blocking).

For analog lines, the ITU recommendations Q.411 through Q.420 define a tone signalling system. However, in some countries, other signalling systems are used instead or additionally (e.g. DC signalling with exchange line feeding or feeding in PABX).

The digital line signals are defined in the ITU recommendations Q.421 through Q.429. These signals are transmitted via the signalling channel, as per G.732 and G.734 (channel-associated signalling). Individual countries can stretch these recommendations to include other signals, e.g. charge signals. The ITU recommendation Q.430 describes how the two signalling systems are converted.

MFC-R2 Line Signalling

In HiPath 4000, boards of the same type are all loaded with the same loadware. The HiPath 4000 achieves the necessary flexibility by subsequently uploading additional, variable parameter blocks from the system to each individual board (e.g. timer values). This allows the HiPath 4000 system to assign different parameter blocks to each individual line/circuit (or channel, in the case of DIU-CAS PCM30 lines) of a board.

The line signals are received and interpreted by the board loadware, and forwarded as messages to the system for supervision purposes (usually to the device handler). Outgoing signals are initiated by messages from the system, which are interpreted by the board loadware and converted into the appropriate physical criteria for analog lines, or bit strings, in the case of PCM30 highways.

The board loadware has system interfaces to the device handler, to the administration and maintenance system (operating software), and the dependability system.

MFC-R2 Register Signalling

The MFC register signals are inband signals exchanged between registers in the originating exchange and the destination exchange in a forced frequency passage system. The signals are divided into forward signals (originating exchange to destination exchange) and backward signals (destination exchange to originating exchange). The MFC-R2 register signals provide the necessary information for setting up connections.

The forward signals consist of the dialling information (dialled digits), type and origin of the connection, etc. Each signal (1...15) consists of a combination of two frequencies (2 out of 6). In order to be able to differentiate between forward and backward signals, two different frequency ranges are used. The forward signals are sent in the upper frequency range (1380...1980 Hz in steps of 120 Hz).

The backward signals consist of the dialling information request, the called line status signals, called line authorisations, etc. The backward signals are transmitted in the lower frequency range (1140...540 Hz, in steps of 120 Hz).

Some PTT administrations only use the R2 signals 1 - 10. This can save on encoding and decoding devices (signals are then combined with 2 out of 5 tones instead of 2 out of 6).

Forced Frequency Passage Signalling according to ITU

In forced frequency passage signalling, each signal from the originating exchange (A) to the remote exchange (B) is transmitted until the appropriate backward acknowledgement signal is received. Signal exchange always begins in the forward direction (signal is offered). As soon as the acknowledging backward signal is detected by exchange A, the transmission of the applied signal is stopped. As soon as exchange B detects the loss of the forward signal, the acknowledgement signal transmission is stopped.

Exchange A then applies the next forward signal in the sequence. The signal transmitted by the source register in exchange A depends on the evaluation of the backward signal received in acknowledgement of the previous forward signal.

The forward signals are divided into Group-I signals and Group-II signals, the backward signals into Group-A and Group-B signals.

2.11.13 Failure Backup Facilities

2.11.13.1 Trunk Failure Transfer, analog trunk line

If the communication server fails, one or more CO trunks are automatically switched through to a specific station terminal.

- Trunk failure transfer on power failure
  - Transfer takes place if the mains or power supply equipment fails.
- Trunk failure transfer on failure of the control system
  - Transfer takes place if a fault occurs in the central control system preventing trunk calls.
- Transfer and re-transfer after elimination of the fault are automatic, but calls are interrupted.
- Transfer can only be made to a station terminal with a signalling method the same as that of the exchange.
- The station terminal to which transfer is made does not have to be connected to a PABX connection element.
- If a station terminal connected to a PABX connection element is used, it is defined by jumpering at the main distribution frame.

2.11.13.2 Automatic Station Release

Connections are released as far as the station line circuit after a certain time in the event of a/b short-circuit or ground at tip.

- If a user has not finished dialling within a certain time, common equipment is released.
  - In internal calls:
    After a timeout set with the administration and maintenance system, followed by busy tone for a limited period.
  - On external lines without end-of-dial criterion:
    After 7 s, followed by call status.
- If a user does not clear down the connection after the end of a call, common equipment is released after a certain time.
  - After the timeout the user receives the busy tone for a limited time (followed by diversion to station line circuit).

- In the consultation hold status, after the unlimited-duration busy tone has started the user can still hang up or cancel the consultation hold or activate features (after 10 seconds of busy tone, the signal key must be pressed first on DTMF telephones).

2.11.13.3 Duplex Operation

In duplex systems the central system components (i.e. principally the switching processor) are duplicated. While one processor is actively handling switching operations the STANDBY processor processes the information supplied via the tie trunk on the connections that have been set up.

When the reasons for changeover are:

- fault in the active processor,
- manual activation, or
- routine activation at defined intervals,

system control switches over from the active processor to the STANDBY processor. The STANDBY processor performs a SOFT RESTART and then becomes the active processor. At the end of a HARD RESTART the defective processor is switched to the STANDBY state.

Any connections which have been set up (between two terminals or trunks) remain intact during processor changeover.

The runtime for a HARD RESTART is a matter of minutes, that for a SOFT RESTART a matter of seconds.

Split operation

To allow service functions such as database regeneration, or generation to be performed without affecting the switching operations, it is possible with the aid of administration and maintenance to change a system over from duplex to simplex mode.

- The active processor retains its status unchanged, but in the event of a fault "soft restart simplex" is carried out instead of "soft restart duplex", in other words the system behaves like a simplex system.
- The non-active processor changes from the STANDBY status to the STAND-ALONE status. A restart of this processor does not affect the active processor.
- When the system data have been regenerated, the system is changed back via administration and maintenance from simplex to duplex mode. The processor that was active up to this point becomes the STANDBY processor, and the STAND-ALONE processor becomes the active processor.

Restart behavior: Soft restart

The relevant computer is reset and proceeds to runup. The code and static data as well as the peripheral units are not reloaded. Soft restart of the redundant units (CC, LTG, CSN) is only possible with changeover (partner control must be in the standby condition). Through-connected two-party connections are retained.

Both soft restart duplex and simplex is possible. Soft restart simplex is carried out whenever the standby half is either not present or is not in the standby condition.

With the current recovery concept, only standard two-party connections in talking state are not interrupted during soft restarts (SRs). These connections are saved according to the "two-party-save" principle (status data of caller and called party).

After recovery, saved connections are in a special call processing state from which they can only be terminated (cleared down). Other call processing functions are no longer possible.

Standard two-party connections are connections in talking state between two communicating parties. This can mean one of the following connection types:

- connections between two internal station users
- connections between a line circuit and an internal station user
- connections between a line circuit and a special function terminal
- connections between two line circuits

as long as no other user or circuit or attendant console is participating.

Connections between a station user or a line circuit and the attendant console are not saved.

The recovery parameters can also be set to "one-party-save" or "call-ref.-save" via AMO. This ensures that exchange connections are saved during recovery as follows:

1. Incoming digital and analog exchange calls, which have already been answered by the called party (exchange has received answering criterion) and are no longer in a standard two-party connection state, are saved according to the "one-party-save" principle, i.e. only the status data of the exchange line is saved.

   These incoming exchange calls are re-rung at the attendant console after the soft restart has been completed.

   Incoming digital and analog exchange calls, which are saved according to the "one-party-save" principle, include the following connection types:

   - Attendant console connections in any state after answering, e.g.
     – Exchange line connections in talking state (exception, since these connections are currently not saved with the "two-party-save" principle)

   – Exchange calls on consultation hold
   – Exchange calls on hold
- Exchange calls on consultation hold
- Exchange calls participating in a three-party conference
- Exchange calls in intrusion (override) state
- Exchange calls with call waiting for internal station user (knocking)
- Parked exchange calls
- Exchange calls on toggle hold

2. Outgoing digital exchange calls, which are not saved according to the "two-party-save" principle, are terminated by the soft restart protocol.
3. Incoming or outgoing digital exchange calls in other call processing states, such as
   - connection setup state
   - connection cleardown state,

   are terminated by the soft restart protocol.
4. The "one-party-save" or "call-ref.-save" recovery only saves exchange connections. Tie-trunk connections are not saved; tie-trunk connections set up via exchange lines in the remote system report the soft restart to the exchange circuit via the message interface.
5. For internal network configurations, digital networking trunks are required for this feature. This is because some circuit release data cannot be transmitted to exchange trunks in foreign nodes via analog lines.

Restart behavior: Hard restart

Following power-on, the computer is reset and proceeds to runup. All checksums are checked and the inconsistent subsystems and the periphery are reloaded.

In all other cases of hard restart, such as

- escalation of soft restarts,
- hard restart LTG simplex,
- sporadic central error,

restartable peripheral modules are not reloaded.

If the hard restart relates to an active control half of the SWU and if changeover with soft restart is not possible, all relevant connections will be released.

1. Reload

   As for hard restart, but all subsystems are always reloaded.

2. Module restart

   The restart of an IEC bus user (SWU or server) does not r

2.11.14 DTMF Tone Suppression

Third-party voice mail and IVR servers that receive in-band DTMF information from the HiPath 4000 system upon answer are configured in HiPath as a Special Outpulsing Device (SOD). Whenever a caller is routed to an SOD, the HiPath 4000's tone generator is used to send DTMF tones to the third-party server that is connected to the SOD interface. These tones are used to convey called/calling party information to the third-party server "in-band", or via the speech highway. These tones are meant to be heard only by the third-party server and not by the calling party.

The DTMF tone suppression feature requires the HiPath 4000 system to attenuate or open the path between the receive side of the SOD interface and the calling party such that the reflected DTMF tones cannot be heard by the caller. In quantifiable terms, while DTMF tones are being sent to the third-party server, no DTMF component shall be delivered to the interface of the calling party at a level higher than -55 dBm.

It is realized that DTMF tone suppression not cause clipping of any audible information (e.g., an IVR voice prompt) that is intended to be heard by the caller after the DTMF tones are sent to the third-party server. To meet this requirement a normal speech path with standard attenuation levels (compliant with HiPath 4000's transmission and loss plan) between the receive side of the SOD interface and the calling party must be restored within 100 milliseconds following the delivery of the last DTMF digit sent to the third-party server.

While the DTMF tones are being sent to the third-party server the calling party is either continue to connect to audible ring back tone or hear silence.

DTMF tone suppression is applied for calls that are routed to an SOD regardless of the location and type of originating device. This includes calls received from a public telephone network, calls received from the private network, and calls originated by users (e.g., anate, digite, or attendant) in the local switch.

The routing of a call to an SOD may occur as a result of any of the following call scenarios:

- A-B call to the SOD
- Transferred call to the SOD
- Forwarded call to the SOD

- Hunting to an SOD that is a member of a pilot number hunt group
- ACD call to an SOD that is configured as an ACD Agent
- A call directed to an SOD by means of a step in an ACD Routing Table (ART)

Restrictions:

DTMF tone suppression is supported on calls that are routed to a third-party server regardless of the originating device type.

DTMF tone suppression is supported for third-party servers that are connected to the HiPath 4000 on an interface configured as a Special Outpulsing Device (SOD).

DTMF tone suppression applies to third-party servers, which rely on in-band DTMF tones to receive information from the HiPath 4000 switch. However, this feature does not impact to the server equipment or their applications whatsoever.

2.11.15 1st Party CTI / X_Messages

X-Messages are used to control 1st party TAPI service providers. The X-messages are part of the CorNet TS protocol which is used to control digital system- and IP-telephones .

The X-Messages allow to transport the following information:

- to the TSP: dynamic, static feature presentation, transmission of status indications, party numbers and names
- to the switch: in addition to the CorNet TS-message X-Containers are used to carry information regarding feature activation

Restrictions

Digital system- or IP-telephones with multi-lines are not supported

2.12 Workstation Protocol Adaptions

All digital system telephones and options are CorNet-TS controlled. From the phones the connection to the PC is via USB-interface. CorNet-WP will be passed through to the connected PC.

The USB class driver for the data stream which is necessary for CTI (AT-commands, Cornet-commands and WP-elements) is tunneled through USB and offered on a virtual COM port. The existing TAPI service-provider (TSP) and the existing CTI applications (if they are compatible to Win 98 and Win 2000) are used.

USB

This module will handle the communications between phone and PC via USB. It will offer the PC the following features of the phone:

- Audio Device Class: This class offers the Audio Features of the phone to the PC. The PC can use the audio features of the phone like a standard sound device.
- Access to the B and D channels for voice and data calls. The USB module simulates an ISDN Adapter, for data calls. If the PC needs workstation protocol or X- containers the USB module has to simulate the control adapter, too.

Workstation Protocol Interface / "Protocol Class" Conzept

The message set allowed to communicate between the Service Provider and Switch will be primarily dependent on the workstation's protocol class.

Each device will use either: CorNet-TS, CorNet-TS / WP or CorNet-TS / WP+.

– CorNet-TS is composed of CorNet-TS messages to communicate with the service provider.

– CorNet-WP is composed of all CorNet-TS messages and a subset of CorNet-WP messages.

– CorNet-WP+
is composed of all CorNet-TS messages as well as all CorNet-WP messages.

The message set associated with each protocol class are defined as follows:

Protocol class: CorNet-TS

WP Messages:

– None

Protocol class: CorNet-WP

WP Mesages

– Call Substate Notification

– Called Party Number

– Calling Party Number

– Connected Party Number

– DNIS Identification

– DNIS Number

– DSS Identification Number

- Dynamic Feature Presentation
- Feature Activation
- Feature Indication
- IE Mask
- Keypad Facility
- Last Call Update
- Line Identification Number
- Program Mode
- Redirecting Party Number
- Redirection Party Number
- Results Code
- Static Feature Presentation

protocol classCorNet-WP+

WP-messages

- All of the messages listed for CorNet-WP, as well as:
- Called Party Name
- Calling Party Name
- Connected Party Name
- DSS Identification Name
- Line Identification Name
- Redirecting Party Name
- Redirection Party Name
- Time and Date key
- Message Waiting Light Restored

Capacities

The number of simultaneous CorNet WP sessions that can be supported is one per phone.

All phones and remote workstations can be configured as workstations to use CorNet WP.

Performance

CPU and SLMO performance limitations with respect to WSP. To avoid expected performance with the HiPath 4000 and CorNet WP 2 classes of Client types are defined:

Basic user

This class covers the simple diallers, and will get no additional features that are provided only within the workstation protocol.

WP-user

This class offers additional TAPI support.
How many WP-users can fit onto the same SLMO/SLMOP is dependent on the size of any pikkupgroups, key systems, the line and feature usage rate, and the requested WP msgs sent between it and the client. The maximum number of WP users that will fit onto the same switch regarding the specific switch processor type.

Audio, Signal, Display IEs

The CorNet-WP messages "Audio", "Signal" and "Display" will not be sent to any HiPath 4000 workstation due to a reduction in messaging traffic.

3 Basic Features

3.1 Connectable telephones

Analog telephones

- Telephone with rotary dialing (pulse dialing)
- Telephone with push-button dialing (pulse dialing)
- Telephone with push-button dialing (DTMF according to ITU Q.23)

Digital system telephones

- digital system telephones
- enhanced operator console
- Certified DSS1 terminals

IP-telephones and softclients

- IP-telephones
- softclients
- enhanced operators console

SIP-telephones

- SIP-telephones

3.1.1 Headset on digital system- and IP-telephones

It is possible to operate the digital system- and IP-telephones using a headset instead of a handset. The headset replaces the handset in such scenarios.

1. Taking a call or entering the digital input state using the "headset" key

   The LED assigned to the "headset" key

   - flashes rapidly when the phone is ringing,
   - remains on for the duration of a call,
   - remains dark when the phone is in an idle state.
2. The "headset" key has a toggle function for taking calls/entering the digital state and subsequent release.

3. When a headset is connected, the hook switch is not active, except where the the loudspeaker key on the terminal has been activated (in this case the loudspeaker is switched off).
4. When the headset is replaced by a handset, the hook switch functions normally again.
5. Hands-free talking is not possible, but the loudspeaker can be switched on to support the headset.
6. The "headset" key can be configured using AMO.

3.1.2 Message Waiting Indication (MWI) in hotel applications

HiPath 4000 supports lamp control on analog terminals. This mode permits cost-effective provision of Message Waiting functionality (especially important in the hotel industry) using standard US devices with lamps (from AT&T, for example). Prerequisite is that HiPath is converted as follows:

- Installation of US ring current generator Q2468 in a modified shelf.
  - Ring current 85V, amplitude 20 Hz
  - MWI signal -108V DC

The implementation complies with the European safety standard EN 60950 and UL 10950/ UL 1459. <u>The approval process for this feature has been carried out for Thailand only.</u> Use in other countries must be arranged for on a case-by-case basis.

- The MWI is activated by iniating a callback on free, in the same way as COMTEL method used in Switzerland. Retrieval is initiated by lifting the handset.
  - The MWI lamp can also be activated via the ACL command. No callback is entered in this scenario.
- Due to use of the US ring current generator, only a limited number of call cadences are possible (default: 2 sec. ring, 2 sec. interval).
- The "MWI" feature can only be activated when US RG and circuit type are correctly configured.
  - COMTEL signaling is set up in the system.
  - When the combination of COMTEL signaling and US RG is detected, the LW in the SLMA converts the COMTEL signaling to US MWI signaling.

3.2 Dialling

3.2.1 On-Hook Dialling

- Using a digital system- or IP-telephone a connection can be established without lifting the handset.
  - The user hears the call progress tones via the loudspeaker of the digital system- or IP-telephone.
  - On a digital system- or IP-telephone with handsfree talking equipment, in addition to the loudspeaker, the microphone is also switched on when the user starts dialling.
- Dialling by pressing the loudspeaker key: the user gets a dial tone on the loudspeaker.
- Dialling without lifting the handset or pressing the loudspeaker key is possible by using the following keys:
  - keypad,
  - repertory key,
  - DSS key,
  - number redial key,
  - voice calling key,
  - loudspeaker key.
- The LED of the loudspeaker key is lit following the dialling as long as the loudspeaker is switched on.
- Switching off the loudspeaker
  - by lifting the handset,
  - by pressing the loudspeaker key.
- The connection is terminated by pressing the loudspeaker key, provided the handset was not lifted.
- If a station user (in idle state) presses a number redial key, a DSS key or the repertory key without stored dialling information, a connection without dialling is carried out.
  - The station user receives an internal dial tone.
  - Subsequently the user can dial, using the keypad or another DSS/ repertory key.

- If the user is dialling without lifting the handset and the call reaches a busy partner of an external installation, which is linked to the HiPath 4000 using an analog interconnection, the user receives, so far, no busy tone; in fact "connect" and subsequently "disconnet" is signaled, that is the HiPath user is set to idle state.

  With HiPath 4000 in this situation the user receives a busy tone giving him the opportunity to:

  – disconnect, using the loudspeaker key,

  – let the connection disconnect itself after some time.

3.2.2 Display when dialling

1. The displays of the digital system- or IP-telephones in a HiPath 4000 system are controlled with the aid of the "flexible display control" feature. With the aid of the ZAND AMO, the display output can bee chosen amongst 4 different modes:

   - The telephone number and name of the caller are shown on the display. The name of the caller is cut if there is insufficient room on the display.
   - The name of the caller is always suppressed, the telephone number is displayed in full length.
   - The telephone number is always suppressed, the name of the caller is displayed in full length.
   - The telephone number and name of the caller are shown, if there is sufficient room on the display. If there is not sufficient room, one of the 2 information types takes precedence, depending on the AMO parameter setting, i.e. either

     – the name or

     – the telephoine number is omitted. In the latter case, the fill-in character "#" is substituted for the telephone number.
   - The 24-section display offers 21 characters for displaying the information, not counting the blanks and special characters.

2. During dialling with the keypad, the digits dialled by hand are written from left to right as they are entered.

   - For internal call numbers, excess digits are ignored and not displayed.
   - For external call numbers, a maximum of 22 digits can be shown.

3. When an internal call number is dialled for which a name is entered in the system, once entry of the call number is completed the name is shown along with the displayed call number.
   - Names are entered using the administration and maintenance system.
   - In addition to names, other information can be entered and displayed (e.g. first names).
   - Names can also be displayed for internal destinations which have been dialled by code (e.g. hunting group, public address system).
4. Displays related to activities of the station user himself (e.g. dialling, input) have priority over passive displays (e.g. name of the caller). As soon as the station user has ended his activity and the existing display is cleared, any other waiting display information from the system is shown.
5. Status displays have priority over call charge displays for this connection.
6. If the call is forwarded as a result of call forwarding - all calls, call forwarding - no answer or hunting group, the name of the called station user and the name of the destination station user are displayed.
7. The display time of the call charges at the end of a call is uniformly defined for the digital system- and IP-telephones by means of AMO.
   - Call charge display (10 s by default)
8. The display time of the call number after answering or dialling the external number is uniformly defined for the digital system- and IP-telephones by means of AMO.
   - Call number (20 s by default)
   - Call number (5 s by default)
9. Display for speed dialling, repertory call and DSS
   - When the feature is used
     – After dialling the speed call number, the display changes to the stored call number. If available, the name is also shown.
     – After the repertory or DSS key is pressed, the call number and the name (if available) are shown.
   - When the feature is interrogated
     – After the speed call number has been dialled, the display changes over to the call number and the name (if available).
     – After the repertory or DSS key has been pressed, the call number and the name (if available) are shown.
   - When the user inputs data

   – After entry of the call number for the DSS key, the name, if available, is shown in addition to the digits.

10. Display during redial

   - For repetition or interrogation of last number redial, the call number and the name (if available, e.g. for internal destination) are displayed after the number redial key is pressed.

3.2.3 Repertory Keys (Function Keys)

Users equipped with digital system- or IP-telephones have repertory keys (function keys) at their disposal.

No local memory is reserved for digital system- and IP-telephones. A defined shift key has been set on the add-on terminal, allowing the user to assign a 2nd level to all repertory keys which have been set on the add-on terminal. This 2nd level doubles the amount of repertory keys on add-on terminals.

If a key on the add-on terminal has been set as a function key, the function remains when the shift key is used. The shift function only effects keys which are set as repertory keys.

Pressing the shift key on one of the connectable add-on terminals activates or deactivates the second level on all other add-on terminals. The corresponding LED indicates the given status: LED on indicates that the 2nd level is active; LED off indicates that the 1st level is active.

1. One internal or external call number can be stored on each repertory key.
2. There is a limit of 22 digits per destination (including the trunk group code for external destinations).
3. The fax or DTE codes can also be stored on repertory keys, it is also possible to enter the appropriate code before using the repertory key.
4. All dialling information entered on the pushbutton set (i.e. including * and #) can be stored.
5. The functions of feature keys (key codes) can also be entered on the repertory keys.
6. Non-relevant keys in the programming mode (TR, MA, etc.): the keystroke is ignored for the programming mode but it does affect any voice connection in existence at the same time.
7. For each terminal there is a central memory for 10 repertory keys. In addition, each terminal can be assigned a number of memory blocks for name keys/ DSS keys (in blocks of 5). The memory for the repertory keys is assigned with the aid of administration and maintenance. The memory blocks can be assigned from a common pool until used up.

8. In the maximum configuration all users can have repertory keys.
9. If the user in the idle state presses an repertory key for which no dialling information is stored, then seizure without dialling takes place.
   - The user receives the internal dial tone.
   - A message appears on the display to indicate that the memory is empty.
   - The user can then dial with the pushbutton set or another feature keys.
10. Pressing a repertory key during a call initiates automatic callback.
11. Manual suffix dialling for external calls
   - After the repertory key has been pressed, further digits may be dialled and these are sent out after the stored call number.
   - Digits suffixed manually in this way are displayed successively immediately after the stored call number.
12. Repertory keys can be used for entering call numbers (check function) (for variable call forwarding, for example).
13. Entering a destination
   - The user can enter a call number or procedure on any repertory key.
   - Any number already stored under the repertory key is overwritten.
   - A number of destinations can be entered one after the other without having to start the procedure from the beginning each time.
   - individual repertory keys can be set for terminals so that they can not be reprogrammed by mistake when using the terminal. These protected keys can only be programmed using AMOs.
14. Deleting a destination

   Any destinations entered by a user can be deleted
     – by overwriting with another destination,
     – with the aid of the delete procedure.
15. Checking the destination
   - A user can check the call numbers stored on the repertory keys by pressing the CH key and then the appropriate repertory key.

3.2.3.1 Rep-Dial (DDS, Name Key)

IMPORTANT: The requirements below apply to station Repdial keys and do not apply to the Attendant Console.

Allows the programming of flash and pause timing (multiple pause timing) into a repdial key by users and administrators.

- Each repertory dial entry may contain up to 22 digits.
- The station user will be able to program a pause into the Repdial digit.
- The pause time associated with each pause command is a System parameter that will be administrable by AMO (the default value will be 3.0 seconds).
- The station user will be able to program a pause command.
- The station user will be able to program a Consult (flash) into the Repdial digit string.
- Each entry may contain the digits 0-9, *, #, and certain feature keys or pause.
- Chaining entries can not be done for storing multiple features within a single REPDIAL key. (E.g., Call Forward access code followed by destination + DND access code).
- A REPDIAL key can contain a single access code followed by the associated information for that access code (e.g., Call forwarding access code + call forward destination).
- Allow DTMF signaling capability when initiating a call from a RepDial key
- When a DDS key with no programming is pressed an implicit flash is generated and the user can then dial a destination or other feature.

3.2.3.2 CONNECT key feature

The new CONNECT combines the functionality of **Toggle** and **Pickup for digital system and IP-telephones**.

- The CONNECT key can be used in the following situations:
  - Answer Camp-on Calls
    Accept a call that has been Camped-on to the users extension. The Camped-on call can be an internal or external call.
  - Alternate between two parties during a Consultation call allows the user to talk privately with a second party while the first party is on soft hold. The Connect feature can be used to alternate between the two parties.

- Reconnect to connections and held (soft hold) parties (e.g. conferences, consultation held party)
  Reconnect with a consultation held party if the user initiates a consultation call and the consulted-to party does not want to take the call, or the consulted-to party is busy, or does not answer.
- Group Pickup
  Accept a pending Group Call Pickup for the users group.

- The Connect key feature is supported on digital system- and IP-telephones.
- One hundred percent of the digital system- and IP-telephones are able to be configured with the CONNECT key.
- All digital system- and IP-telephones, configured with the CONNECT key, are able to have access to the Connect key feature.
- The Connect key feature can be used when active on primary, secondary, and phantom lines.
- The CONNECT key can be made part of a standard key layout by administration.

Restrictions

- There is no access code for the Connect key feature.
- CMI (wireless telephone) can not be configured with a CONNECT key.
- The telephone user can not configure the CONNECT key.

Feature Interaction

Display features

- The LED associated with the CONNECT key flashes indicating a call is pending.
- The Connect function is not supported in the Optiguide menu system.
- Using the CONNECT key while in preview mode deactivates preview mode.

Holding features

- Consultation / Flash (Consultation Hold)

  When active in a consultation call, the CONNECT key can be used for toggling, but not for group call pickup.
- Automatic Recall on Held Calls (Recall)

When a station user is active in a connection and a recall occurs to the line that the user is active on the recall camp-on to the line does NOT take place. Therefore, there is no interaction with the Connect key feature and recalls.

This applies to:

– recall from Park to Station.

– recall from Park to System.

– recall from Group Park.

Call Origination

- The Connect key feature can be used when active on any line where multiline appearances are configured.

Night Arrangements

A digital system- or IP-telephone, configured as a night station, can use the Connect key feature.

3.2.4 Prepared Dialling

It is possible to enter call numbers and to modify them before dialing.

The digits are collected firstly but not yet evaluated by the system.

It is possible

- to delete digits in the selected string,
- to insert digits or
- to complete the string.

Afterwards the dial is executed as en-block dialing, as the OK-key or the LS-key is pressed or the handset is lifted. The feature is realized for digital system- and IP-telephones.

Key	Function
Loudspeaker key`-`	Backspace
Menu key `<` `>`	navigate left, right

Key	Function
Menu key „Ok“` Lift handset Loudspeaker key Line selection	Activate dialing, start of dialing the entered number
Service Menu key	Cancel dialing

Restrictions

- Collecting digits is only possible in idle state (handset on-hook, no dial-tone).
- The digits are displayed in the first row of the display.
- It is possible to navigate within the string with the '<', '>' keys.
- Menu navigation is not possible!
- Dialed digits are inserted at the cursor's position.
- An overwriting of digits is not realized.
- It is possible to delete digits at any position, by pressing the loudspeaker-key '-' . The '-' Key acts as a backspace-key. Always the digit left of the cursor gets deleted.
- The delete-key deletes the whole dial string and it is possible to start a dial again.
- By pressing the Last-number-redial-key, name-key or direct-station-selection-key, the stored digits are copied into the display. The previous digits are deleted.
- The enter-mode can be canceled by pressing the service-key or the release-key.
- The primeline is used for keyset.
- At the keyset, the dial on this line can be executed by pressing this line in the edit mode. This action can be on the primeline or a secondary line.

3.2.5 Outgoing Traffic

3.2.5.1 Setting Up Outgoing External Calls

1. Seizure of external lines (trunks and tie trunks) by
   - dialling the trunk group code (or dial CO/exchange code, e.g. "9")
   - speed dialling

- number redial
- repertory keys
- If fax or modem data lines are dialled on analog lines the call number must be prefixed by the appropriate service identifier.

2. Trunk dial tone is applied by the system when the CO trunk is seized. Customer-specific music or an announcement can be applied instead of this tone. A separate device connection is provided for this.
3. In order not to seize central equipment (such as code receivers) unnecessarily, a timeout period is set after seizure. If the user does not dial before this timeout period expires the central equipment is released.
   - Analog telephone:

     Timeout of approx. 10 s, then 10 s busy tone or an appropriate announcement (3.5 s), followed by forcible release.
   - Digital system- or IP-telephone:

     Timeout period of approx. 10 s, then busy tone for 10 s and a display message (5 s) then lockout on the station line.
   - Dial tone is sent for a maximum of 10 s.
   - In the consultation state lockout does not take place (unlimited busy tone).
4. For lines without an end-of-selection criterion a timeout period of 10 s is set after each digit dialled via the external line; if the user does not dial another digit before the timeout period expires, the system assumes the call state (artificial end of selection).
5. The external party releases the connection.
   - External connection without release criterion: possible busy tone from the line
   - External connection with release criterion
     – User makes a call with the handset:
       Busy tone for 10 s then call holding state
     – User makes a call with the handsfree conversing equipment (digital system- or IP-telephones): terminal is in the idle state; LS LED is off; a new call can be signalled.

3.2.5.2 Setting Up Outgoing Internal Calls

1. Setting up an internal call
   - dialling the call number,
   - by speed dialling,
   - by number redial,
   - by repertory keys,
   - by direct station selection keys,
   - by the hot-line service.
   - If fax or modem data lines are dialled on analog lines the call number must be prefixed by the appropriate service identifier.
2. If all the code receivers are busy the user can wait off- hook until he receives internal dial tone.
3. Setting up a connection to an Infobox to leave a message.
4. In order not to seize central equipment (such as code receivers) unnecessarily, a timeout period is set after seizure and after each digit dialled. If the user does not complete the dialling before this timeout period expires the central equipment is released.
   - Analog telephone:

     Timeout of approx. 10 s, then 10 s busy tone or an appropriate announcement (3.5 s), followed by forcible release.
   - digital system- or IP-telephone:

     Timeout period of approx. 10 s, then busy tone for 10 s or a display message (5 s) then lockout on the station line.
   - Dial tone is sent for a maximum of 10 s.
   - In the consultation state lockout does not take place (unlimited busy tone).
5. A caller who accesses a busy trunk group to the voice mail system receives a negative acknowledgment. This case is treated in the same way as all trunks busy. Override, camp-on and entry of a callback request are not possible.
6. The internal party releases the connection.
   - User makes a call with the handset: Busy tone for 10 s then call holding state

- User makes a call with the handsfree conversing equipment (digital system- and IP-telephone): terminal is in the idle state; speaker LED is off; a new call can be signalled.

3.3 Switching Functions

3.3.1 Consultation Hold

In the case of digital system- and IP-telephones, a special key is used to initiate and cancel consultation as well as to pick up calls and to toggle. With analog telephones, this function can be performed

- with the grounding button,
- with the flash key, or
- by dialling a single-digit code.

Consultation hold is also possible throughout an integrated network, via a second line.

3.3.1.1 Consultation Hold during External Call

1. Consultation hold during an outgoing or incoming external call for an internal or external user or the attendant
   - Consultation hold during a trunk call
   - Consultation hold during a tie trunk call
   - Consultation hold to an external user via a trunk
   - Consultation hold to an external user via a tie trunk
   - Consultation hold to an external user via a satellite PABX line
   - Consultation hold to an internal user
   - Consultation hold to the attendant console
2. Users of analog terminals cannot initiate a further consultation call if they are already engaged in one.
3. Users of digital system- and IP-telephones who already are the consulted party can initiate another consultation call. However, the maximum number of consultation calls which can be linked is two.
4. In the consultation state the consulting user can use the following additional features:

- Override/camp-on
- Callback
- Paging
- Conference
- Hold toggle
- Storage of dialling information

5. On pushbutton telephones the features are activated in the consultation state by pressing the signal key.
   - If the feature is activated within the timeout period after the last digit of the call number selected for consultation hold has been dialled, the signal key does not have to be pressed.
   - The signal key does not have to be pressed if feature keys are used, which are combined with an integrated flash
6. If features are to be activated on an analog telephone in the consultation state, the code must be dialled within 2 s of pressing the signal key otherwise the system will cancel the consultation hold.
7. Any features for which there is no corresponding feature key can be selected on a digital system- and IP-telephone in the consultation state without pressing any key equivalent to the signal key.
8. After the consultation state has been initiated (by pressing SIG, RF, NA etc.), the existing call is placed on hold (splitting) if the call permits consultation hold (e.g. initial call, second call transferred with AUN).
9. A user on hold hears music, or an announcement, or nothing until the consulting user returns to the call.
10. As an alternative to the standard music options, the waiting party can be supplied with a melody or announcement selected by the customer.
11. Cancelling consultation hold is possible
    - before the user answers
    - after the user answers.
    - In the case of an analog terminal, the original call is established only after a timeout period of 2 s (features can be activated before the timeout period expires).
    - If the user on hold releases the connection before consultation hold is cancelled the consulting user hears the busy tone after consultation hold is cancelled.

12. Transfer of a further call by the consulting user during consultation hold (digital system- and IP-telephones) by pressing the AUN key.
    - Return to consultation call
      - with disconnection of the transferred call
      - with hold toggle. The toggle-key is used to alternate between the consultation call and the transferred call.
    - Transferring the transferred call.

      Go into consultation hold from the transferred call, dial the call number, press the UEG key in the new consultation call.
13. No consultation from voice mail connections.

3.3.1.2 Consultation Hold during Internal Call

1. From an initial call both parties can go into consultation hold alternately. If the user on hold also tries to go into consultation hold this attempt is ignored.
2. A user placed on hold through consultation or call transfer cannot go into consultation hold but a user with a digital system- or IP telephone can accept a call in this state.
3. All users can enter the consultation state from a connection to the attendant console. Consultation is not possible from a console-console connection (extending).
   A user connected to a console can:
   - initiate consultation
   - cancel consultation
   - alternate between console and consultation party
   - transfer in the following states:
     - ringing
     - call
     - busy

   The consulted user can
   - take the call

   During consultation by a user the following possibilities are available to the console:
   - splitting/error: the connection to the consulting party is released, the connection between the consulting and the consulted parties remains;

- call transfer when the consulted party has answered.

During consultation by a user the following possibilities are **not** available to the console:

- park/hold (with the release key or by changing the call key),
- transfer in the ringing or busy state.

In the consultation state it is possible to

- transfer/pick up
  – The name and call number of the consulted party are displayed at the console.
  – If the consulted party was accessed by means of call forwarding, the name and call number of the forwarded party (not the consulted party) are displayed at the console.

The consulting party is recalled in the following cases:

- after a specific time if the consulted party does not answer following transfer in the ringing/busy state
- immediately upon cradling the handset if transfer is not possible.

4. No consultation fromvoice mail connections.

3.3.1.3 Station Transfer Security

- If the consulting user goes on-hook in the consultation state and this does not lead to a call connection being established between the user on hold and the user consulted, an automatic recall is initiated for the call on hold.
- Immediate automatic recall for the consulting user
  – if a call is transferred to an unavailable station line,
  – if a call number has been dialled which does not exist,
  – if all trunks are busy,
  – if a call is transferred to a station line which is out of service,
  – if the station terminal to which the call is transferred does not have the class-of-service corresponding to the call on hold,
  – if there is already a call waiting for the user to whom the call is transferred,
  – if the user to whom the call is to be transferred has not yet pressed his signal key in systems with call transfer through call pickup,
  – if the user attempts to transfer a trunk call to an external party,

  - if going on-hook during consultation means that the call is to be transferred to a user with the "do-not-disturb" feature activated (unless the consulting user is authorised to violate the "do-not-disturb" feature),
  - if the user has not dialled or not dialled the full number in the consultation state.
  - Timed recall to the consulting user
  - The time after which automatic recall is to be signalled can be defined with the aid of administration and maintenance.
  - Automatic recall takes place if the free user to whom the call is to be transferred does not answer within the timeout period.
  - Automatic recall takes place if the busy user to whom the call is to be transferred does not become free and answer within the timeout period.
  - If for automatic recall for an external call the consulting user is free and does not answer before the timeout period expires again, the attendant console receives the automatic recall.
  - If the consulting user is busy during automatic recall of an external call, the attendant console immediately receives the automatic recall.
  - If for automatic recall for an internal call the consulting user is free he receives ringing tone until he answers or the waiting user goes on-hook.
  - If for automatic recall for an internal call the consulting user is busy the recall timeout period is restarted again and again until the consulting user or the waiting user goes on-hook.
  - Automatic recall has priority over call forwarding - no answer at the station terminal to which the call is to be transferred.
- Automatic recalls intended for the attendant console are signalled at the night station during night service.
- If an external call is not answered within a recall time it is released. The timers can be adjusted with the aid of administration and maintenance.
- If call forwarding - all calls, call forwarding - no answer, hunting group or "do-not-disturb" is activated for the station terminal of the consulting user these features are ignored for the automatic recall.
- An automatic recall cannot be picked up from a different station terminal.
- If a parked call is not picked up within the recall timeout period, the station terminal at which the call is parked is sent an automatic recall.
- If the toggling user withdraws from a toggle connection (goes on-hook) he receives an automatic recall from the party of the initial call if the procedure does not lead to call transfer.

- During the automatic recall the waiting user hears the following:
  - Music, or the "Hold the line please" announcement, or nothing
  - Digital system- or IP telephone / attendant terminal: display (for a maximum of 5 s in the case of digital system- or IP telephone)
  - As an alternative to the standard music options, the waiting party can be supplied with a melody or announcement selected by the customer.
  - Recall after on hook
  - After toggling in 1st consultation (B--A--C) this feature allows the consulted to party 'C' to recall the consulting party 'A' if the following conditions are met:
    - 1. The consulting party 'A' (digital system- or IP telephone) has toggled back to the formerly held party 'B'
    - 2. The party 'B' has to go on-hook now
    - 3. Afterwards the consulting party 'A' goes on-hook

    This causes the remaining and still waiting party 'C' to recall the party 'A'.

    If the feature is not active, the call is dropped and no recall happens. Without the feature being active a recall can only be from 'B' to 'A' but never from 'C' to 'A'.

3.3.1.4 Manual Hold of Consultation

An digital system- or IP-telephone user who receives a consultation call from another local or networkwide station (via CorNet NQ) can put the call on Manual Hold/Private Hold of Consultation in order to perform other functions, such as originate a call on another line key. Neither the originator of the consultation call nor the user who is on Consultation Hold can use Manual Hold.

Features

The feature "Manual Hold of Consultation" will allow an user as the consulted party, to place the call on manual hold. This works for first and second consultation. The functionality is available for manual hold and private hold (=exclusive hold).

The consulted user is not allowed to have a background connection at the same time. In this case manual hold is not possible.

1. **Manual Hold of first consultation**

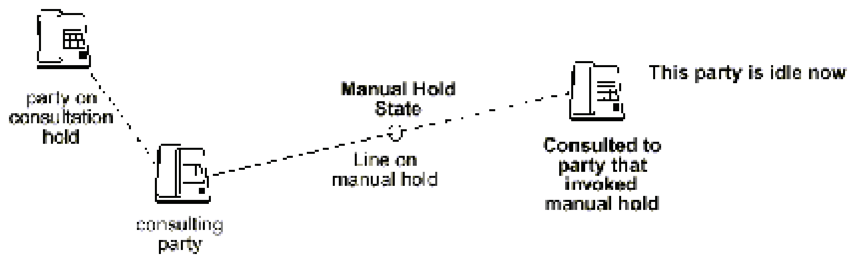


Figure 20 Activation of Manual Hold of 1st Consultation:

2. **Manual Hold of second consultation**

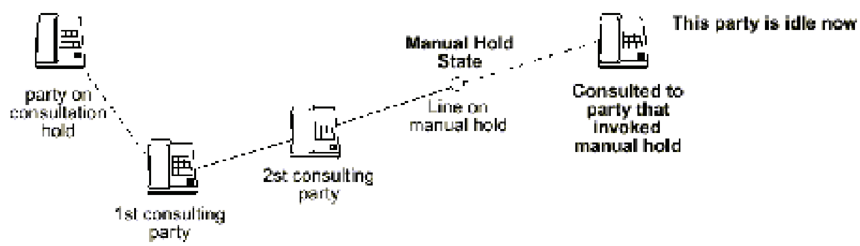


Figure 21 Activation of Manual Hold of 2nd Consultation:

Interaction with other features:

– Manual Hold of Consultation call can be used by ACD agents.

– The originator of the consultation call can transfer the call during the time the consulted user has the call on Manual Hold.

– During the time that the call is on Manual Hold, the originator of the consultation (consulting party) is not able to establish a conference.

– Automatic Recall on Held Calls applies. The manual hold recall timer is not reset upon transfer of the call by the consultation call originator.

– Music on Hold (if equipped) is provided to the held user.

– It is possible to do a Directed Call Pickup of a line which has a consultation call on Manual Hold.

– The originator of the consultation call is able to toggle between the user who is on Consultation Hold and the consulted party who has activated Manual Hold.

3.3.2 Extending Trunk Calls to Attendant Console/ Night Station

1. Transfer of external calls to the attendant console without prior console answering
   - The consulting user goes on-hook after dialling the answering code (in systems with normal call transfer)
   - During night service the call is signalled at the night station.
2. Call transfer to attendant
   - Users can transfer a call on hold to the attendant after dialling the answering code in the consultation state. This can take place before or after the attendant answers (transfer before the attendant answers is treated in the same way as going on-hook during consultation).
   - The call is signalled on the call key which corresponds to the call to be transferred (if the user on hold is an external party the call is signalled on the A key).
   - When a call transferred to the attendant in this way is answered, the attendant is connected in the call state to the consulting user.
   - The user can toggle between the user on hold and the attendant.
   - The user can cancel the consultation call to the attendant terminal.
   - After transfer by the consulting user in the call state with the attendant, the attendant is connected to the user on hold. Signalling changes to display the original call on hold.
   - If the consulting user in a call transfer to the attendant goes on-hook before or after the attendant answers, the system reacts as follows if the attendant has not already picked up the waiting call:
     – System with transfer through pickup:
     Automatic recall to the consulting user
     – System with normal transfer:
     Transfer of the waiting call to the attendant
3. Saving exchange calls during switchover to night operation

   If a headset is disconnected from the attendant console (automatic switchover to night operation) during a call, the connection is automatically switched to another attendant or to the night service station.

3.3.3 Holding a Call

1. Holding a consultation call

   Pressing the following keys causes the call in progress to be held, and the station user initiating consultation can set up a new call

   - consultation key
   - repertory key
   - DSS key
   - voice calling key

2. Holding a call after announcement of a call

   - During a call, a call can be announced by means of visual camp-on or override. The desired station user can put the original call on hold. The announced call is then automatically switched through.
   - Holding is possible with the following features
     - override
     - camp-on
     - call waiting – terminating
     - call pickup
     - direct call

3. Holding a call during alternating between parties

   - If he has a first call and a second call, the station user can alternate between the two, i.e. each call is either in the call state or put on hold.
   - Alternating between calls after
     - consultation,
     - pickup of a call.

4. If the station user who has put a call on hold goes on-hook, he receives a recall.

3.4 Changeover to Night Service

3.4.1 Special Night Answer Position

The calls intended for the night service are signalled at one or more night stations during the night service condition.

1. Night service for calls on external lines
   - trunks
   - tie trunks
2. Night service for attendant calls
3. Calls on the P key of the attendant terminal are not switched to the night service (this also applies to external calls on the P key).
4. Types of night station
   - Normal night station (analog / digital system- or IP telephone)
   - Attendant terminal which is operated in alternation with the attendant terminal of the normal attendant console (night attendant terminal)
   - Night stations and night attendant terminals can be used together (in shared systems).
5. Assignment of night attendant terminals to attendant groups
   - Night attendant terminals can each be assigned as a night service variant to an attendant group.
6. A night attendant terminal performs the following functions:
   - Selection and activation of another night service variant
   - Class-of-service changeover is signalled to the normal attendant console when changeover to daytime operation takes place.
   - Activation/deactivation of controlled group restriction
   - Controlled station-to-station restriction
   - Alarm signalling
   - All "personal" features (e.g. programming repertory keys)
7. No more than one special night answer position may be active for each attendant group.
8. A group comprising any of the stations may be entered with the aid of administration and maintenance as a special night answer position.
   This group comprises

- a maximum of 10 stations,
- call distribution,
- console answering priorities, as defined for the special night answer position.
- If external calls switched to night service are waiting for the night station to become free the station terminal next in line for a call according to the call distribution rules receives an alerting tone.
- Automatic class-of-service changeover for all stations in the group, as defined for the special night answer position.

9. With the aid of administration and maintenance a night service variant can be entered for which signalling of the calls switched for night service is suppressed.
10. A recorded announcement equipment or answering service as the night station is possible.
11. Console answering priorities for calls at the night station
    - Answering in order of arrival
    - Console answering priorities according to type of traffic and type of call
        – Priorities
          Emergency call
          Trunk calls - automatic recall
          Trunk calls - initial call, announcement traffic - external
          Tie trunks - preferred trunk groups
          Tie trunk calls - initial calls and recalls, attendant calls, announcement traffic - internal
        – The sequence of priorities can be changed with the aid of administration and maintenance. (The sequence of console answering priorities is identical for the attendant console and the night station).
12. Notification to the night station that external calls switched to the night service are waiting for the night station to become free.
    - Night station with analog telephones: alerting tone
    - Night station with digital system- or IP telephones:
        – The call that has been waiting longest in the call queuer is automatically extended to the backup memory of the night station if this is free. By giving priority to external calls these are signalled in preference to the user as secondary calls (visual call waiting indication)

- The calls switched to night service automatically camp on irrespective of whether the night station has activated "camp-on enable" or not.
- When it has been entered in the backup memory the call leaves the call queue.

- Night attendant terminal: the priority alert lamp lights (as in daytime operation)
- Alerting tone is sent to the night station in the initial call and to the night station on consultation hold.
- While alerting tone is being sent to it the night station can go on consultation hold and transfer.

13. Automatic class-of-service changeover for the night station in night service
    - Classes-of-service assigned to a night station in night service
      - Override
      - Transfer (applies only to analog telephones; digital system- or IP telephones can always transfer)
      - Trunk access (local, area, toll). This class-of-service can be defined for all the stations of a system with the aid of administration and maintenance.
    - If a night station has activated transfer of ringing, calls switched to night service are forwarded but the classes-of-service of the destination station terminal are not automatically changed over as in the case of the night station.
14. Extending incoming calls by the night station
    - analog telephone / digital system- or IP telephone: by transfer
      - Transfer is also possible if the destination station has switched to the voice mail by activating call forwarding/transfer of ringing or is a member of a hunting group with overflow to the voice mail activated.
    - Night attendant terminal: as for the normal attendant console
15. Extending outgoing calls by the night station
    - Extending/assigning undialled lines
      - The class-of-service of the user who requires the call is switched over to the uniform class-of-service defined in the system for the duration of this call (as in the case of extending/assigning by the attendant).
    - Extending/assigning dialled lines
    - analog telephone / digital system- or IP telephone: by transfer

- Night attendant terminal: as for the normal attendant console

16. Camp-on during call extending
    - analog telephone / digital system- or IP telephone: by transfer
    - Night attendant terminal: as for the normal attendant console
17. Automatic recall during call extending
    - analog telephone / digital system- or IP telephone: as for going on-hook during consultation
    - Night attendant terminal: as for the normal attendant console
18. Entering/cancelling/interrogating the night service variant, selecting the night service variant, activating/deactivating night service
19. Automatic night service
20. Saving established exchange connections when activating the night option. If an attendant unplugs the headset during an ongoing connection (i.e. activates the night option), the connection is switched to another attendant or to the night service terminal.

3.4.2 Universal Night Answer

The calls intended for the universal night answer are signalled in parallel at one or more shared ringing units during the night service condition.

1. The calls can be answered or extended from any station terminal which has a class-of-service appropriate to the call.
2. Universal night answer for calls on external lines
    - trunks
    - tie trunks
3. Universal night answer for internal calls at the attendant console (no personal calls)
4. Normal audible bells can be installed as the ringing units.
5. Console answering priorities for calls with universal night answer
    - Answering in order of arrival
    - Console answering priorities according to type of traffic and type of call
        – Priorities
          Timed hot-line service
          Trunk calls - automatic recall
          Trunk calls - initial call, announcement traffic - external

Tie trunks - preferred trunk groups, tie trunk recall, call from a preferred user, mobile radio call
Tie trunk calls - initial calls and recalls, attendant calls, announcement traffic - internal

– The sequence of priorities can be changed with the aid of administration and maintenance. (The sequence of console answering priorities is identical for the attendant console and universal night answer).

6. Negative acknowledgment if the call has already been answered by another user or if the answering user does not have the class-of-service appropriate to the call.
7. Extending after universal night answer by transfer
8. Attendant camp-on during extension of calls after universal night answer through call transfer
9. Automatic recall during call extending after universal night answer as in the case of going on-hook during consultation.
   Automatic recalls which are normally signalled at the attendant console are signalled with the aid of the shared ringing units and can be answered from any station.
10. Entering/cancelling/interrogating, selecting, activating/ deactivating the night service variant "universal night answer".

3.4.3 Permanent Night Station

HiPath 4000 can operate without an attendant console.The call queue is initialised as if it were in 'day service operation',but will be switched to the set night variant with the first call after system cutover if there is no attendant console. If there is an attendant console, but the jack has been withdrawn or the attendant console has not been plugged, changeover to the night variant is effected on system cutover.

3.4.4 Remote Activation of Night Station

All authorised terminals within the same system group network can define night switch options, and therefore also switch the night service allocation to a different terminal, if required.

The terminal types permitted as authorised terminals are analog telephones(DTMF dialling; otherwise pulse-dialling of codes defined in the central speed dialling list SPDC), digital system- and IP telephones and attendant consoles. Night service stations are entered by AMO in the NOPT list ("night variant" / night option list) assigned to each group.

The night service option desired is selected by means of the following procedure:

- Dial the code for activating of the function
- Dial the answering code of the group whose NOPT list you wish to access, and the node in which the night service terminal is configured and
- Enter the night variant number to select the desired night service option.

These codes can only be dialled from an authorised terminal or attendant console. The code combinations may also be programmed onto the repertory keys of the terminal concerned.

The night option changeover (i.e. activation of new night option and deactivation of old night option) can be carried out depending on the state of the attendant console. If one attendant console is still in day operation, only one presetting is active.

The night option changeover can be carried out by any authorised user or attendant.

Each dialled code is positively or negatively acknowledged by the system, depending on whether the desired changeover is acceptable or not.

Within their home node, authorised users can activate the same night service option for up to 3 attendant console groups in one "session"; authorised users dialling in from other nodes can only activate the night option for one attendant console group in the same session.

If the night service attendant is absent, the night service terminal's own "night option" will become active. This switches all calls to the voice mail service for the duration of the attendant's absence. To this end, all terminals defined as night service options must be assigned the feature "call forwarding on no answer", with a short "no answer" timeout (1s) active only during night operation. The timer must revert to its standard setting during day operation, and the fixed destination entered must be the voice mail.

3.5 User Guidance

3.5.1 Optiguide User Interface

Functions

The Main Menu key

The Main Menu key is a new additional key on digital system- and IP telephones and Key Modules. Therefore a key function (MAIN MENU) is available for every key map template and every subscriber key map. Only one Main Menu key is allowed per station. The existing Program/Service key can coexist on a digital device configured with the Main Menu key. The Main Menu key is not user programmable. There are two slightly different types of main menu: "main menu key idle menu" in idle state and "main menu key non-idle menu" in non idle states.

A system option decides which menu has to be displayed when "<" key or ">" key is pressed in idle state. The default of this system option is "dialog idle menu". Main Menu is terminated by time out or by pressing the Main Menu key again. The display time has a value of 1 to 254 seconds. The default value is 5 seconds.

Program/Service Mode

Administration offers a new parameter to administer the new system option "no service mode during voice call". This system option restricts the user from entering service mode while non-idle. This system option is deactivated by default.

Last Number Redial (LNR) / Saved Number Redial (SNR)

Additional feature prompts in dialog menu for Last Number Redial (LNR) and Saved Number Redial (SNR) during dial states (e.g. initial dial state --> immediate display, consultation dial state --> via scrolling). For SNR menu prompts in other states are added to save the number.

Consult only

This feature allows a station user to "consult only" with a party without fear of accidently transferring the held party to the consulted-to party. By going on-hook after selecting the consult menu item no transfer will occur. User shall be recalled from the held party. Transfer is possible by pressing the transfer key or by menu. The consult only feature is activated by a system option. By default, it is deactivated.

Camp on

The Optiguide feature prompt (dialog prompt) allows the user, of a digital system- or IP telephone, to answer any camped-on call. The prompt is supported for automatic or manual camped-on, second calls, and delayed call forwarding on busy features.

Camp on protection

Camp-on to a digital system- or IP telephone is blocked if the digital telephone is classmarked as "camp-on protected"

Main menu structure in idle state within HiPath 4000

New Menu items in main menu idle state	submenus
Speed dial features	Last number redial (also in dialog menu)
	Station speed dial
	Saved number redial (also in dialog menu)
	System speed dial 1
	System speed dial 2
	Previous menu
More features	Enter account code
	Call forwarding
	Do not disturb (for feature phones and keysets)
	Ringer cutoff (for keysets only)
	Previous menu
View active features	Show only active features for
	Call forwarding
	Do not disturb
	Hunt group (only if subscriber is hunt group member)
	Speaker call reject
	Second call
	Ringer cutoff
	Callbacks pending --> go to service mode
Call log	Incoming calls
	Outgoing calls
Program/Service	--> go to service mode
Phone settings	--> go to service mode / phone settings
Exit	--> go to idle display (e.g. time and date)

Main menu structure in Non-idle state

Menu items in main menu NON- idle state	submenus
Speed dial features	Last number redial (also offered in dialog menu)
	Station speed dial
	Save number redial (also offered in dialog menu)

Menu items in main menu NON- idle state	submenus
	System speed dial 1
	System speed dial 2
	Previous menu
More features	Enter account code
	Do not disturb (for feature phones and keysets)
	Ringer cutoff (for keysets only)
	Open listen/Speaker
	Show used line
	Previous menu
View active features	Show only active features for
	Call forwarding
	Do not disturb (for feature phones and keysets)
	Hunt group
	Speaker call reject
	Second call
	Ringer cutoff (for keysets only)
	Callbacks pending --> go to service mode if system option allows
Call log	If service mode system option allows:
	Incoming calls
	Outgoing calls
Program/Service	--> go to service mode if system option allows
Exit	--> show non-idle display (e.g. for talk state)

3.5.2 User Signalling

Feature User Signalling consists of 5 parts; Tones, Ringing, LED cadences, DTMF DNIS SID/ANI displays and Dialling / DTMF Signalling.

Tones

Call progress tones that are presented to the user are in general country specific. In HiPath 4000 the tone types are configurable for frequencies, cadences and level. This information is a function of the Loadware, and it is also possible to change this information by AMO (customer specific). Tone tables are implemented for country adaptations (i.e. country specific).

Ringing

Programmable ringing types are located in a database. HiPath 4000 administration flexibly sets ringing cadences for all station types (ring cadences for US ring types are hardcoded) . Additionally the ringing types are configurable for for different ringing situations in call processing.

LEDs

The 5th. flash rate (flutter) are provided in HiPath 4000. ,Winking (a medium cadence) is used for a held line and a ringing line flashes (slow cadence). Cadences are programmable for ringing and hold.

DNIS SID/ANI

HiPath 4000 supports DNIS and SID/ANI displays for ACD and plain station users and provides the DNIS, SID/ANI display even after through connect. Repeat ID-key is available to allow DNIS, SID/ANI information to be brought back to the display.

DTMF Dialling and DTMF Signalling

In HiPath 4000, DTMF-signalling will be invoked on demand of the called station. Once required, it will be automatically activated in a call. Automatic DTMF-signalling will be provided in conference connections and in incoming and outgoing external connections. Local conference members have the option to get blocked from hearing DTMF-tones. When DTMF signalling is activated each keypad dialed number will be converted into DTMF signalling. Impacted are the following existing features when activated via feature code during a basic talk state or in consultation talk state :

– Account Code and Malicious Call Trace
– Delete ->Conference Member Display Information (during conference only)
– Call Test Tone/ Call Frequency Test (France)

When automatic DTMF is activated on demand, the digits stored at the dialing feature key will be converted into DTMF-signals and sent to the demanding partner. When the partner doesn't require DTMF (no demand), the digits has to be subjected the digit evaluation by the originating party.

Restriction

The feature User Signalling contains no US-specific requirements for display indications.

3.5.3 Facility for Short Announcements

3.5.3.1 Recorded Announcement for Waiting Party

1. During a connection in which a call state does not exist, an external or internal station user/attendant receives audible notification of the held state.
   - Music
     – 2 melodies are stored.
     – One melody can be selected using an administration and maintenance operation.
     – As an alternative to the standard music options, the waiting party can be supplied with a melody selected by the customer.
   - Announcement, standard ("Please hold the line" in 13 languages).
   - The administration and maintenance system can be used to select between music or an announcement.
   - Audible notification of held state can be suppressed.
   - Continuous repetition of the music or announcement
   - Two or more waiting parties can hear the music or announcement at the same time.
   - For tenant service the music or announcement is the same for all customers.
   - When an external notification source is used (music or announcement), the "equipment for connecting individual melodies or individual announcements for the waiting party" is required.
2. States in which the waiting party hears music or an announcement:
   - A connection is held at the attendant console (automatic hold on the called key, parking on key).
   - During call switching or during consultation.
   - In a call queue for a busy station after the call has been switched or transferred.
   - During automatic camp-on for a busy station (for which call waiting - terminating has been activated) after the call has been switched or transferred.
   - During call waiting for a busy attendant console after transfer and for a busy hunting group station after the call has been switched or transferred.

- In the ringing condition for a free station after the call has been switched or transferred.
- During recall at the attendant console or station.
- In hold status because of call pickup.
- In hold status during alternating between calls.
- While the call is parked (from the station).

3. Multiple music on hold

3.5.3.2 Voice Prompting

1. With the help of short announcements, analog telephones are advised of the cause of negative acknowledgments. For example:
   Announcement "This function is currently not possible" following an attempt to enter a callback request when the callback storage "pool" is full.
   - The short announcements are permanent and integrated in the system.
   - A total of 6 announcement texts are available
     – Please hold the line
     – The party you have called is not available
     – This function is not implemented
     – Wrong access authorisation
     – This function is currently not possible
     – You have a message
2. The short announcements are available in different languages.

3.5.4 Open Listening

During a call, a station user with a digital system- or IP telephone can switch on the loudspeaker to enable other persons in the room to listen in on the call.

- The user can continue to listen and speak via his handset during open listening.
- Open listening requires that the terminal has a telephone speaker or handsfree talking equipment.
- Open listening is also possible in the conference condition.

- The LED of the loudspeaker key is lit for as long as the loudspeaker is switched on.
- It is also possible for the loudspeaker to be switched on automatically for a call (voice calling).

3.5.5 Handsfree Talking

Digital system- or IP telephones can be equipped with handsfree talking equipment.

- Listening with handsfree talking equipment switched on via built-in loudspeaker.
- Speaking with handsfree talking equipment switched on via the built-in microphone.
- The volume is set on the terminal. Depending on the level of the call, the volume of all calls is regulated by the automatic level equaliser (within certain limits) to the set value.
- Handsfree talking equipment is activated by the following keys when the handset is on-hook:
  - keypad,
  - repertory key,
  - DSS key,
  - number redial key,
  - voice calling key,
  - loudspeaker key,
  - ST key (for consultation and voice mail output)
- Handsfree talking is also possible in the conference condition.
- If the handset is lifted during handsfree talking, the handsfree talking equipment is switched off.
- The LED of the loudspeaker key is lit as long as the loudspeaker is switched on.

3.5.6 Display

3.5.6.1 Time and Date Display

On the digital system- or IP telephones the time and date can be shown on the display in any state upon request by the station user.

- The time/date display is cancelled automatically after 5 sec. or by activities of the user.
- Additionally there is a continuous time and date display on the digital system- or IP telephones when in the idle condition.
- The time and date are provided by the switching facility; it is not necessary for the station user to set them.
- The system clock continues to run in the event of a power failure.
- Display on the digital system- and IP telephones.
  - The display of hours and minutes is separated by a colon.
  - Any display already being shown is interrupted for the duration of the date/time display, as long as such displays have a lower priority.
- The administration and maintenance system is used to define the abbreviations used for days of the week (different languages).
- Display of date and time on the attendant console:
  - The display appears in line 12 of the call processing and programming layout,
  - the display appears as it does on digital system- and IP telephones.
- If there are digital system- and IP telephones with freely programmable keys, each of these keys can be set up by the user as a "time key". This key function must otherwise be established by administration and maintenance.

Clock Synchronisation

A clock component with a backup battery supports the display of the time and date for HiPath 4000 systems. This clock component can be set using AMO’s ("DATE" for date and time, SONUS for daylight savings and standard time). Clock accuracy depends on the component’s specific environment (1 sec. per day to 1 sec. per week).

3.5.6.2 Different Time Zones (DTZ)

The feature " Different Time Zones is used in situations where Access Points or IP phones are registered to one host system, but located in different time zones. In that case it is possible to display the local time of day at the digital system phones or IP phones that are connected to the remote Access Point.

- The local time is supported by the following functions:
  - Date and time of the day in the idle display
  - Daylight savings time and standard time
  - Call log
  - Callbacks
- Activated Call Back are stored with the system time into the call back data pool. The callback list is always displayed with the local time
  - Date/time button
  - Date change at midnight
- The display time on the phones is assigned by configurable time classes
- In the HiPath 4000 you can set up to 50 time classes. If time class is set to 0, the local time is equal to the system time.

  Example: Daylight Saving Time DST
- The daylight saving time / standard time switchover can be performed automatically. Due to the fact that DST does not exist in each time zone and DST starts at different dates, it can be defined for each time class when to switchover to Daylight Saving Time or back to standard time.

Restrictions

- Not supported for IP phones as mobile users
- The feature "time reminder" for phones with time class 0 is not supported and will be blocked
- Error messages are generated with system time
- Call Detail Recording is done in system time
- The local time is only used for display features
- The system time is used for routing, triggering, tracing etc.

3.5.6.3 Call Display

The origin of the call is indicated on the display along with call signalling.

1. The call can also arrive during a first call or second call. The display is then accompanied by alerting tone.
2. Displays resulting from an activity by the station user himself (e.g. input, interrogation, dialling) have priority over displays for the call.
   - The call is initially only audibly signalled.
   - The display for the call does not appear until the station user interrupts or terminates his activity.
3. Calls are shown on the display with the call number and name (internally if available) of the calling party.
   - The names are entered using the administration and maintenance system.
   - In addition to names, further information, such as organisational information, can be entered and displayed as long as memory space is available.
4. The displays of the digital system- and IP telephones in a HiPath 4000 system are controlled with the aid of the "flexible display control" feature. With the aid of the ZAND AMO, 4 different modes can be selected for the display output:
   - The calling line information and name of the caller are output on the display. The call number information takes precedence, i.e. the name of the caller is truncated if there is insufficient room on the display.
   - The name of the caller is always suppressed, the calling line information is output in full.
   - The calling line information is always suppressed, the name of the caller is output in full.
   - The calling line information and name of the caller are output, if there is sufficient room on the display. If there is not sufficient room, one of the 2 information types takes precedence, depending on the AMO parameter setting, i.e. either
     – the name or
     – the calling line information is omitted. In the latter case, the fill-in character "#" is substituted for the call number.

     The 24-section display offers 21 characters for displaying the information, not counting the blanks and special characters.
5. Display for calls in which the system knows neither the name nor call number of the calling party.
   - Outside call: <EXT>
   - Tie-line call: <TIE (trunk group number)>

6. Suppression of call number display at the called party if an "unlisted call number" has been set up for the calling party.
   In this case the 5-character symbol information '*XXX*' is displayed.
7. Names or call numbers are shown left–justified.
8. Permanent texts for the displays are available in different languages. Permanent texts can be generated in five of the available languages for each system. You can find a list of the languages in Section 3.5.7, “Individual Language Selection”.

3.5.6.4 Repeat Identification (Repeat ID)

When the user is active and initiates another feature or function, the display is modified and does not contain the original calling or called party information. During the call state by pressing the Repeat ID key the called/connected party information is now displayed. The feature is deactivated by display timeout (e.g. 8 seconds, adminstrable Timer) or overwritten by other user requested or features. Initialization of the telephone also deactivates Repeat ID Feature.

Features

- The Repeat ID key is an optional feature key that can be assigned to any Keyset or digital system- or IP telephone with a display. The station user can press this key while in talk state to display calling and/or called party information.
- The Repeat ID key can be used in the following special call states:
  – Normal Two-party Talk (through-connected)
  – Consultation Talk state for the party initiating the consultation
  – Consulted party
  – ACD silent monitoring, target monitor state and partner state
- The Repeat ID key can be used in connections containing one or more of the following parties:
  – Local (internal station)
  – Private Network (e.g., CorNet NQ)
  – Public (e.g., CO)
  – ACD Agent/Supervisor
  – Special Device
- This feature can be used when the keyset user is active on the primary, secondary, or phantom lines.

- The Repeat ID display is a two-line display.
- A Keyset or digital system- or IP telephone configured as a night station can use the Repeat ID feature
- The feature Repeat ID can only be activated by pressing the Repeat ID key.
- If the Repeat ID key is pressed in a not allowed situation, depression of the key is rejected with a display for 2 seconds
  – -(Line 1) Repeat ID
  – -(Line 2) Not possible

Restrictions

– An analog telephone user does not have access to the Repeat ID feature.

– A Profiset does not support a Repeat ID key.

– Attendant Console does not support a Repeat ID key.

– The Repeat ID key LED is never illuminated.

– A station can not initiate the Repeat ID feature while being held.

Feature Interaction and restrictions

– Message Waiting/Mailbox/Phone Mail Indication / Volume Up / Volume Down
The use of function keys while Repeat ID display is active deactivates the Repeat ID display.

– Push Button Dialing deactivates Repeat ID display if active.

– Priority of Displays on Keyset or digital system- or IP telephones

– When the Keyset or DFT initiates another feature, the Repeat ID display is overwritten with information based on the users' request and display priority for the requested feature

– SID/ANI/DNIS information, if available, is displayed upon initiating Repeat ID while active on a line.

– There are no menu or submenu items in Common User Interface (CUI)for the Repeat ID feature.

– The display information for Repeat ID overwrites the charge display.

– The display information for the Call Log display overwrites Repeat ID information.

Privacy features

Using the Ringer Cutoff key while Repeat ID display is active deactivates the repeat ID display in response to the user action.
Using the Privacy key while Repeat ID display is active deactivates the repeat ID display in response to the user action.

Holding features

A terminal can not initiate the Repeat ID feature while Group Parked.

Transfer features

Call Transfer with Automatic Camp-on

During camp-on both users, the station user that has been transferred and is camped on, and the station user that is "camped on to", can not initiate Repeat ID.

Conference features

Repeat ID can not be initiated while active in a conference or in a consultation conference state.

Busy Line features

- During camp-on the station user that has been transferred and is camped on, and the station user that is camped on to, currently can not initiate Repeat ID.
- During busy override station users in the connection, currently can not initiate.
- During Emergency override station users in the connection, cannot initiate Repeat ID.
- A Second Call information is overwritten by connected party information when Repeat ID is requested for the active line.

Other features and functions

- PROGRAM/SET (Service Mode)
  Using the Program key while the Repeat ID display is active deactivates the Repeat ID display in response to the users' request.
- Repeat ID cannot be used to display information associated with a Night Answer call before the call has been answered.
- Repeat ID is not applicable to receipt of an AICS call.

Networking

Once a call is established with via

  – Primary Rate
  – tie line,
  – DID (Direct Inward Dialing)

– DIT (Direct Inward Termination)
– LCR (Least Cost Routing)
– Central Office (CO)

the Repeat ID feature can be initiated for certain call states (e.g., talk, consultation,).

3.5.6.5 Charge Display

The station can be assigned the class-of-service for call charge indication at station by means of the administration and maintenance system. After the class-of-service has been entered, the station user is automatically shown the call charge for outgoing calls.

Displays resulting from activities of the station user himself (e.g. dialling) or from call signalling (e.g. call waiting) will interrupt the call charge display.

Display in the event of more than one outgoing call per station.
There can be a maximum of 2 calls (primary connection/secondary connection), each comprising a first call and consultation call, on the digital system- and IP telephone. The display shows the combined charges of the respective primary connection (comprising the first call and consultation call). There is therefore one counter each for the primary connection and the secondary connection.

With HiPath 4000, the call charge display function for

- internal
- network and
- incoming exchange

connections is shown in the display of digital system- and IP telephones as a sum of all the foreground connections.

3.5.6.6 Elapsed Time Display (ETD)

When an internal user, with a digital system- or IP telphone with display, makes or receives a call on an external trunk, upon reaching the connected talk state, their telephone will display the cumulative elapsed time of the call.

The cumulative elapsed time refers to the elapsed time to be displayed with respect to the time when the trunk is answered (incoming trunks), or upon answer or simulated answer (outgoing trunks).

Feature Operation

1. The ETD feature is automatically activated if the device is configured for that feature.
2. Provide a station based option to automatically display Elapsed Time, Charge Display or No display. One of three Display options can be assigned:
   – Elapsed Time display
   – Charge Display
   – None (No Elapsed Time or Charge Display)
3. The ETD feature is applicable for calls involving a "supported trunk" call. Herein, the supported trunk consists of the following types of public and private connections. When describing call transactions, the term "Party B" will always refer to the external user on a supported trunk.
   a) CO, DID, ISDN, or other public trunk facility calls.
   b) CorNet NQ trunk calls with classmark status indicating an external call. The classmark status indicates a B-channel connection to an external trunk. No elapsed time display (ETD) will be given if the classmark status indicates that the call is internal.
   c) Other TIE trunk calls.
   d) Other International trunk types
4. The ETD function is subject to the following:
   a) The elapsed time of a call will only be displayed on a call in the connected talk, consultation talk, consultation conference talk, or toggle talk states.
   b) ETD according to the following format:

      HH:MM:SS where HH = Hours, MM = Minutes, SS = Seconds

      Valid examples are: 00:09:59, 1:00:00, 00:00:51.
5. The elapsed time information will appear on Party A's display, as described in the following basic outgoing and incoming call examples.

   Outgoing Basic Call Example:
   As Party A dials Party B, the dialed number is displayed on the top line of the phone display. ETD will be displayed after 10 seconds (or time based on system administration of display timer) after the call enters the connected talk state.

   Incoming Basic Call Example:
   When Party B calls Party A, Party A will see the calling party information (i.e. trunk number, or ANI name/number) on the top line of the phone display. ETD will be displayed after 10 seconds (or time based on system administration of display timer) after the call enters the connected talk state.

6. One hundred percent of supported telephones can use this feature.

Restrictions

– ETD is not supported on CMI telephones.
– ETD is not supported on functional devices.
– There is no access code for the feature.
– ETD will not be given during dialing.
– The ETD feature is not applicable at the attendant consoles.
– Workstation protocol and direct attached first party APIs, e.g., TAPI, will not support this feature.
– The timestamp cannot be transported to another node for the following scenarios:
  – Pickup of a call in another node via directed call pickup
  – Retrieval from group park across the network

Detailed feature interaction

Timed Reminder Service

If the Reminder Timer pops and the user is active on a call the reminder camps-on to the call and display is updated. ETD will be redisplayed after the user answers the reminder call. The ETD will be redisplayed upon exiting progam/service mode after programming of a timed reminder.

Volume Up / Volume Down

The ETD will be redisplayed upon deactivation of this feature.

Priority of Displays

In general: When the digital system or IP telephone is displaying Elapsed Time initiating another function will preempt the existing display. Upon deactivation or timeout of other displays, the ETD will be re-displayed.

Displays During Calling and Repeat ID

During a call, other information may be displayed on Party A's (station) phone, which may replace the ETD. Examples include activating the Repeat ID feature, and "Not Possible" messages. Upon expiration of these displays (after 5 seconds or time based on system administration of display timer), ETD will be given

Common User Interface (CUI)

There is no menu funktion.

Holding

Consultation/Flash (Consultation Hold)
ETD will be given on Party A's phone 10 seconds (or time based on system administration of display timer) after Party A enters the consultation talk state with Party C.

Connect/Brokers/Toggl

10 seconds (or time based on system administration of display timer) after Party A and Party B enter the toggle talk state ETD will be displayed.

Call Park - System

The display of Party C will show the ETD after 10 seconds (or time based on system administration of display timer). The ETD will be calculated to the time when Party B was parked, that is, reset to zero when parked.

Automatic Recall on Held Calls (Recall)

Manual Hold Recall
Upon answering the recall, the answering party's phone will show the ETD after 10 seconds (or time based on system administration of display timer). The ETD will be calculated to the time when the trunk first entered the answered state (incoming trunk), or upon answer or simulated answer state (outgoing trunk). (Not the Held Time)

System Park / Park to station Recall
The answering party's phone will show the ETD after 10 seconds (or time based on system administration of display timer). The ETD will be calculated to the time when the trunk was parked, that is, reset to zero when parked.

Transfer (Call Transfer)
ETD will be given on Party C's phone 10 seconds (or time based on system administration of display timer) after the Party C enters the talk state (in the two-party B-C call). ETD will be calculated to the time of transfer to Party C, that is, reset to zero upon transfer.

Call forwarding

Call Forwarding - General
ETD will be given on Party C's phone after 10 seconds (or time based on system administration of display timer). ETD will be calculated from the time when Party C's phone enters the answered state. For example, ETD will not include the ring-no-answer (RNA) period of Party A on RNA-forwarding.

Conference

When a party is added to a two-party (A-B) connection to form a conference, internal parties using supported phones will have their elapsed time replaced with conference display information.

a) If the conference is reduced to the two-party (A-B) (Station to Trunk) connection, ETD will be displayed on Party A's (Station) phone after 10 seconds (or time based on system administration of display timer). ETD will be reset to zero when the conference is reduced to a two party connection prior to displaying the elapsed time.

b) If the conference is reduced to a two-party (B-C) (Trunk to Station) connection and Party C is an internal supported phone, ETD will be displayed on Party C's phone after 10 seconds (or time based on system administration of display timer). ETD will be reset to zero when the conference is reduced to a two party connection prior to displaying the elapsed time.

c) If Party A (Station) is toggling between a conference and an external consultation party, ETD will be displayed on Party A's phone 10 seconds (or time based on system administration of display timer) after Party A is in a consultation talk state with the external consultation party.

Camp-On

Upon answering, ETD will be given after 10 seconds (or time based on system administration of display timer).

Call Pickup

ETD will be given on Party A's phone after 10 seconds (or time based on system administration of display timer). ETD will be calculated just as if the originally dialed party answered the call. That is to say, upon initiating group call pickup the Elapsed Time is not reset.

Pilot Number Access

ETD will be given on Party C's phone after 10 seconds (or time based on system administration of display timer). ETD will be calculated from the time when Party C's phone enters the answered state. For example, ETD will not include the ring-no-answer (RNA) period of Party A on RNA-hunting.

AUTOMATIC CALL DISTRIBUTION (ACD)

Upon time-out and clearing of an ACD display on the top line, the station user will receive an updated ETD. ETD for a supported trunk in an ACD queue will be calculated with respect to when the trunk is answered.

ACD may route a call by transfer to extension, transfer to agent, or transfer off site. For transfer off site, the Cumulative-ETD information cannot be passed across CorNet. For this case, the time will be initialized in the remote node.

Networking

CorNet NQ
When a call containing a trunk is taken over by a station in a remote node, via the CorNet network, the Elapsed Time information will not be transported to the node where the station took over (retrieved) the call. Therefore, ETD will start at zero upon retrieval or recall of a party from a remote node across the CorNet.

The following functions are not supported:

- Privacy Features
  Data Line Security

- Busy Line Features
  Outgoing Call Queuing - Standby
- Subscriber Hunting Arrangements
  Station Hunt Groups
- Intercom Features
  Executive Intercom (Intercom - Dedicated Intercom)
- Networking
  DISA (Direct Inward System Access)
- Call Origination
  Bridged Call

3.5.7 Individual Language Selection

With HiPath 4000 users can individually select the language of the display texts on digital system- and IP telephones with display.

Available are five languages which are defined during configuration. They are valid per HiPath 4000 System. You can select from the following languages:

- american english
- brazilian
- catalan
- chinese (only digital system telephones)
- croatian
- czech
- danish
- dutch
- english
- estonian
- finnish
- french
- german
- greek
- hungarian
- indonesian
- italian
- japanese (only digital system telephones)
- latvian
- lithuanian
- malayan
- norwegian
- polish
- portugese
- romanian
- russian (cyrillic characters)
- russian (latin characters)
- slovakian
- slovenian
- spanish
- swedish
- thai
- turkish
-

The language is selected either by using AMO ("preferred language") or at the digital system- or IP telephone via the language selection key. The language can also be selected by subscribers with ID card.

If more than one language can be chosen, pressing the language selection key in any state causes the user interface to switch to the next available language. The selected language is displayed for a limited period only and in the required language, i.e. 'Sprache: Deutsch', 'Language: English' or 'Langue: Francais'. As soon as the language change takes place all display texts are shown in this language.

Language selection on a station can be activated and deactivated for certain user groups using COS .

With regard to the duration of the selection, the following modes are possible depending on the COS:

- Temporary selection
  The language selection is valid until the end of the next cal
- Static selection
  The selected language is stored until the next language selection by a user or administrator.
- Language transfer with PIN identification
  If a PIN is used for identification on a different station in either the home node or a foreign node, the language selected on the home station is used for the time of identification. If this language is not available the language set on the foreign station is used.

3.5.7.1 Representation of display texts

The representation of display texts is subject to the following rules:

All fixed display texts controlled by the HiPath 4000 as well as all user interface menu texts are displayed in the selected language.

Variable texts such as subscriber names are displayed as configured in the system or as transmitted within the network.

Announcements generated by the System (e.g. ‘Please hold the line‘, ‘You are not authorized‘) are not injected in the selected language but in the default language configured in the communication server(depending on the SIU).

3.5.7.2 Interdependencies with other features

- Autoset Relocate (digital system telephones)

– The preferred language is an element of the static station data and will therefore be considered by Autoset Relocate.

– In logged-off state the language cannot be switched from the station.

- COS changeover

  The operating mode for language selection on the station remains unaffected by the COS changeover.

- Key layout programming

  The language selection key must be configured by the administration.

- PIN

  The configuration of various PIN types in the Class of PIN defines, whether or not own station data transfer is allowed . This parameter also defines the transfer of the preferred language.

- HiPath ProCenter Entry

  This feature is supported by stations configured as agents in HiPath 4000 Pro Center.

- HiPath cordless

  The comfortable mobile phones Gigaset , used for cordless HiPath 4000, support a selection from five (in international markets seven) languages for the user interface. This function is implemented in the local station. The SLC interface provides 16 translation tables for the generation of the menu texts and a limited number of fixed display texts for these languages. If a language is selected on the local station, the new language is transmitted.

  To avoid interactions between the *cordless* function and the multi-lingual HiPath 4000 user interface, the mobile phones are always configured with the systems default language. They cannot alter the language.

- HiPath Trading and Trading E

  To avoid interactions between the user interface generated locally on the SLMT or SLMY board and the multi-lingual HiPath 4000 user interface, the stations configured for the Trading and Trading E feature are always configured with the systems default language. They cannot alter the language.

3.5.8 Buzz

This feature allows a user to alert (buzz) a pre-defined Keyset or digital system- or IP telephone by pressing the BUZZ key or by keying an access code. Voice communication is not provided.

- The buzzed and Buzzing Keyset or digital system- or IP telephone may be in the busy, idle, Ringer Cutoff or DND state.
- The Buzz sound only be signaled at the prime device (not devices with secondary line appearances).
- Buzz causes the target Keyset or digital system- or IP telephone to be alerted and the originators station number to be displayed for two seconds. Buzz interrupts any other alerting of the pre-defined destination for the length of the buzz (350ms on + 150 ms off = 500ms).
- Multiple devices (e.g., executives) can use the same terminating destination (e.g., assistant) (Keyset or digital system- or IP telephone).
- The Buzz target (destination) must be located in the same node as the originator.
- The originating user can press the BUZZ key multiple times to produce user agreed upon buzz patterns (e.g., one buzz, two buzzes).
- Buzz interrupts any other alerting displays of the pre-defined destination for the length of the display.
- The Buzz ring volume is configurable by the user via Audio settings - "Alert tone volume" control.
- A station configured in a night arrangement can be buzzed and can initiate the buzz feature.

General restrictions

- The Buzz feature is not applicable to analog telephones, functional device, or attendant consoles.
- Only one buzz target can be assigned to the originating device.

Feature Interaction and restrictions

Tones, Ringing, Announcements and Displays

- Tones (Distinctive Ringing)
  Buzz temporarily interrupts ringing at the buzz destination.
- Display treatment when initiating or terminating the Buzz feature temporarily overwrites existing displays on the Keyset or DFT.
- The Buzz feature temporarily overwrites existing displays on the buzz destination Keyset or DFT.
- Buzz key while in preview mode deactivates preview mode
- Buzz requires display of the Buzz feature key.

Privacy features

- Pressing the BUZZ key or keying the Buzz access code causes the buzz target Keyset or digital system- or IP telephone to be alerted even when it is in the DND state or when ringer cutoff is active.

Holding features

Consultation/Flash (Consultation Hold)/

a) Since there is no voice connection when initiating the Buzz feature, interaction with this feature is not applicable from the originator.

b) A station user can buzz a destination from "consultation talk" "consultation hold"and state.

Forwarding features

- Call Forwarding

  An attempt to buzz a destination does not follow call forwarding.

- Delayed Call Forwarding on Busy/Buzz

  a) A station that has a pending camp-on (DCFOB call) can initiate the buzz feature and can be buzzed.

  b) A station user can buzz a destination from "camp-on" state and can be buzzed.

Busy Line features

- Camp-On (Internal Call Queuing - Standby)

  a) A busy destination that has a pending camp-on can initiate the buzz feature.

  b) A station user can buzz a destination from "camp-on" state.

- Second Call/Buzz

  a) A busy destination that has a pending second call (camp-on) can initiate the buzz feature and can be buzzed.

  b) A station user can buzz a destination from "second call ringback" state and can be buzzed.

Abbreviated dialing features

- The access code for Buzz can be configured/programmed and saved into a Save/Repeat key.
- The access code for Buzz can be configured into System/Station Speed.

Keyset functions

- The Buzz feature is associated with the prime line of the device. The Buzz target must be the prime line of a Keyset.
- The I-use display is overwritten.
- A phantom line can not be configured as a buzz destination/target.
- Buzz interrupts any other alerting of the pre-defined destination (Audible Ringing on Rollover Lines).

Intercom features

- Broadcast
  a) A Keyset, digital system- or IP telephone does not receive Buzz alerting during broadcast until all users are open in broadcast mode. Upon all speakers becoming active (open) the buzz originator and buzz destination is signaled.
  b) After establishing a two-way connection the buzz feature can be initiated.

Integrated executive / secretary features

Ringing Tones shall be temporarily interrupted by Buzz.

Other features and functions

- PROGRAM/SET (Service Mode)
  a) Buzz alerts a target destination even if the destination is active in program mode.
  b) Buzz can be initiated when the Keyset, digital system- or IP telephone is active in program mode.
  c) Buzz can not terminate to a station active in Audio test or audio setting mode.

d) A station active in audio test or audio setting mode can not initiate buzz.

- A user can not configure the BUZZ key to a Keyset or DFT.
- Automatic Set Relocate: A "signed-off" station can not initiate Buzz and can not be buzzed.

Devices and Terminals

Attendant

An attendant can not be a buzz destination (target), but can initiate the buzz feature.

3.5.9 Intercom Features

Features provided:

– Speaker Call - One-way
Provides the capability to allow users of analog terminals and digital system- or IP telephones to initiate a speaker call, which provides a one-way connection to a single destination (prime line) of their choice.

– Speaker Call - Fixed One-way
Provides the possibility to store the feature into a DDS-key. The DDS-key contains the connections index number and its destination.

– COM Group Call
Provides the prossibility to initiate calls from digital system- or IP telephones within a work group and to establish a two-party connection by using only few digts. The connection establishment is based on the used terminal, not on the used line.

– COM Group Call - Fixed
Provides the possibility to store the feature into a DDS-key. The DDS-key contains the connections index number and its destination.

– COM Group Speaker Call - Two-way
Provides the possibility for users of digital system- or IP telephones to call members within the same COM-group, to automatically activate the targets speaker, and to talk.
To establish the connection, the user pushes the intercom-key or dials the access cose and the targets COM-Group number. Microphone and speaker (provided existent) of the target terminal are activated automatically .

- COM Group Speaker Call - Fixed Two-way (uses COM Group Speaker Call - Two-way feature programmed onto a DDS key)
  Provides the possibility to store the feature into a DDS-key. The DDS-key contains the connections index number and its destination.
- Speaker Call - One-way - Broadcast
  Provides the capability to initiate a speaker call to multiple (maximum 40 keysets) keyset destinations simultaneously. Upon auser answering the broadcast, that user is connected with the originator of the broadcast. All other connections are deactivated.
- Intercom features can be accessed from idle, dial, talk, and consultation dial states.

3.5.9.1 Intercom features and terminals

Speaker Call - One-way

- A Speaker Call - One-way terminates at a digital system- or IP telephone the progam/check mode and is turned into an A-B call.
- A Station user cannot configure a Speaker Call - One-way function key on their telephone.
- The REJECTION key can be used to close a Speaker Call connection.
- CDR/SMDR will treat a Speaker Call - One-way as a basic internal call.
- When active in Baby Listening, this feature can not be accessed.
- A telephone configured as a night station can use this feature.

Analog Telephone

- A Speaker Call will not terminate to an analog telephone. An attempt will be converted to a standard A-B call.

Digital system- or IP telephone

- A Speaker Call - One-way can terminate to a digital system- or IP telephone.
- A Speaker Call can be initiated from a digital system- or IP telephone.

Off Premise Station(OPS)

- A Speaker Call can be initiated from an analog telephone.
- A Speaker Call can not be initiated from a S0-telephone (functional) .

Profiset (Functional Telephone)

- A station user cannot terminate a Speaker Call - One-way to a Profiset. An attempt will be converted to a standard A-B call.

- A Profiset cannot initiate this feature.

Others

- The Attendant does not have access to the Speaker Call -- One-way feature.
- A station user cannot initiate a Speaker Call across the CorNet.NQ.
- A station user cannot initiate a Speaker Call across a trunk of any kind (TIE, DID, CO, etc.).

COM Group Speaker Call - Two-way:

- A COM Group Speaker Call destination telephone cannot be located in a remote node. The call is converted to an A-B call.
- A Speaker Call - One-way terminates at a digital system- or IP telephone the progam/check mode and is turned into an A-B call.
- A Station user cannot configure a COM Group Speaker Call - Two-way key on their telephone
- The REJECTION key can be used to close a Speaker Call connection.
- CDR/SMDR will treat a COM Group Speaker Call - Two-way as a basic internal call.
- When active in Baby Listening, this feature can not be accessed.
- A telephone configured as a night station can use this feature.

Analog Telephone

- An analog telephone cannot be a member of a COM Group. Therefore, a COM Group Speaker Call - Two-way cannot terminate to an analog telephone
- A COM Group Speaker Call - Two-way cannot be initiated from an analog telephone.

Digital system- or IP telephone

- A Speaker Call can terminate to a digital system- or IP telephone.
- A Speaker Call can be initiated from a digital system- or IP telephone.

Off Premise Station (OPS)

- An analog or S0 (functional) telephone cannot be a member of a COM Group.
- A COM Group Speaker Call - Two-way cannot be initiated from an analog telephone.
- A COM Group Speaker Call - Two-way cannot be initiated from a S0-telephone (functional).

Profiset (Functional Telephone)

- A station user cannot terminate a COM Group Speaker Call - Two-way to a Profiset.
- A Profiset cannot initiate this feature.

Others

- An attendant has no access to the COM Group Speaker Call feature.
- A station user cannot initiate a Speaker Call across a trunk of any kind (DID, CO, etc.).
- A station user cannot initiate a Speaker Call across the CorNet.NQ.

COM Group Call

- A COM Group call can terminate to a digital system- or IP telephone in program/check mode.
- A Station user cannot configure a COM Group Call key on their telephone.
- The REJECTION key can be used to close a COM Group Call connection.
- CDR/SMDR will treat a COM Group Call as a basic internal call.
- When active in Baby Listening, this feature can not be accessed.
- A telephone configured as a night station can use this feature.

Analog Telephone

- An analog telephone cannot be a member of a COM Group.
- A COM Group Call cannot be initiated from an analog telephone.

Digital system- or IP telephone

- A digital system- or IP telephone can be a member of a COM Group.
- A COM Group Call can be initiated from a digital system- or IP telephone.

Off Premise Station (OPS)

- An analog or S0 (functional) telephone cannot be a member of a COM Group.
- A COM Group Call cannot be initiated from an analog or S0 telephone.

Profiset (Functional Telephone)

- A station user cannot terminate a COM Group call to a Profiset.
- A Profiset cannot initiate this feature.

Others

- An attendant has no access to the COM Group Speaker Call feature.
- A station user cannot initiate a COM Group Call over CorNet.NQ.

- A station user cannot initiate a COM Group Call across a trunk of any kind.(TIE, DID, , etc.).
- COM Group calls via DISA is not supported.

Speaker Call - One-way - Broadcast

- Speaker Call - One-way - Broadcast cannot terminate to a digital system- or IP telephone in program/check mode.
- A Station user cannot configure a Speaker Call - One-way - Broadcast key on their telephone.
- The REJECTION key can be used to close a Broadcast connection.
- CDR/SMDR will treat a Speaker Call - One-way - Broadcast as a basic internal call.
- When active in Baby Listening, this feature can not be accessed.
- A telephone configured as a night station can use this feature.

Analog Telephone

- A Speaker Call - One-way - Broadcast will not terminate to an analog telephone.
- A Speaker Call - One-way - Broadcast can be initiated from an analog telephone.

Digital system- or IP telephone

- A Speaker Call - One-way - Broadcast can terminate to a digital system- or IP telephone and is converted to a Speaker Call - One-way call.
- A Speaker Call - One-way - Broadcast can be initiated from a digital system- or IP telephone.

Off Premise Station (OPS)

- A Speaker Call - One-way - Broadcast will not terminate to an analog or S0 (functional) telephone.
- A Speaker Call - One-way - Broadcast can be initiated from an analog or S0 (functional) telephone.

Profiset (Functional Telephone)

- A station user cannot terminate a Speaker Call - One-way - Broadcast to a Profiset.
- A Profiset cannot initiate this feature.

Others

- The Attendant does not have access to the Speaker Call -- One-way - Broadcast feature.
- A station user cannot initiate a Speaker Call across a trunk of any kind (DID, CO, etc.).
- A station user cannot initiate a Speaker Call over CorNet.NQ.

3.5.9.2 Restrictions

- Analog telephones cannot be members of a COM Group, so it is not possible to make a COM Group call to an analog telephone.
- Certain Intercom functions are not available to analog telephones, digital system- or IP telephones, or attendant consoles. All functions apply to keysets.
- An attempt to initiate a Broadcast or COM Group call, to a telephone that is not a digital system- or IP telephone or Keyset will result in the initiation of a standard A-B call.
- While active in call log (service mode) intercom features cannot be accessed.

3.5.9.3 Interaction with other features

DTMF Dialing

These functions cannot be accessed via DTMF dialing:

- Speaker Call - Fixed One-way
- COM Group Speaker Call - Two-way
- COM Group Speaker Call - Fixed Two-way
- COM Group Call

Rotary Dialing

These functions cannot be accessed via rotary dialing :

- Speaker Call - Fixed One-way
- COM Group Speaker Call - Two-way
- COM Group Speaker Call - Fixed Two-way
- COM Group Call

Display Features

- It is not possible for a station user in a Speaker Call - One-way - Broadcast connection to initiate the Repeat ID feature.

Holding Features - Consultation/Flash (Consultation Hold)

- Speaker Call - One-way
  - A Speaker Call -- One-way cannot be placed on consultation hold.
  - A digital system- or IP telephone or keyset user on consultation hold cannot initiate a Speaker Call - One-way.
- COM Group Speaker Call - Two-way
  - A COM Group Speaker Call -- Two-way cannot be placed on consultation hold.
  - A digital system- or IP telephone or keyset user on consultation hold cannot initiate a COM Group Speaker Call -Two-way.
- COM Group Call

  A digital system- or IP telephone or keyset user on consultation hold cannot initiate a COM Group Call.
- Voice Calling (Speaker Call - Two-way)
  - A Speaker Call -- Two-way cannot be placed on consultation hold.
  - A digital system- or IP telephone or keyset user on consultation hold cannot initiate a Speaker Call.
- Speaker Call - One-way - Broadcast
  - A Speaker Call -- One-way - Broadcast cannot be placed on consultation hold.
  - A digital system- or IP telephone or keyset user on consultation hold cannot initiate a Speaker Call - One-way -Broadcast.
- Connect/Toggle Key
  - Toggling with another party is not possible.
  - If the user presses the CONNECT key while beeing in a consultation call ; the user will receive NOT POSSIBLE.
- Speaker Call - Fixed One-way
  - Toggling with another party is not possible.

  - If the user presses the CONNECT key while beeing in a consultation call ; the user will receive NOT POSSIBLE.
- COM Group Speaker Call - Two-way
  - Toggling with another party is not possible.
  - If the user presses the CONNECT key while beeing in a consultation call ; the user will receive NOT POSSIBLE.
- COM Group Speaker Call - Fixed Two-way
  - Toggling with another party is not possible.
  - If the user presses the CONNECT key while beeing in a consultation call ; the user will receive NOT POSSIBLE.
- Speaker Call - Two-way
  - Toggling with another party is not possible.
  - If the user presses the CONNECT key while beeing in a consultation call ; the user will receive NOT POSSIBLE.
- Speaker Call - One-way - Broadcast
  - Toggling with another party is not possible.
  - A user can access features associated with Brokers/Toggle once a two party connection is established.
  - A user can initiate this function, if a connection was established because of a consulation call.

Call Hold

- Speaker Call - One-way
  - A held party cannot initiate the Speaker Call feature.
  - A Speaker Call -- One-way cannot be placed on hold (call hold).
- Speaker Call - Fixed One-way
  - A held party cannot initiate the Speaker Call feature.
  - A Speaker Call -- One-way cannot be placed on hold (call hold).
- Voice Calling (Speaker Call - Two-way)
  - A held party cannot initiate the Speaker Call feature.
  - A Speaker Call -- Two-way cannot be placed on hold (call hold).
- Speaker Call - One-way - Broadcast

- A held party cannot initiate the Speaker Call feature..
- A Speaker Call - One-way - Broadcast cannot be placed on hold (call hold).
- Once a Speaker Call - One-way - Broadcast becomes a two party connection, this connection can be placed on hold.

Manual Hold

Speaker Calls cannot be placed on manual hold.

Transfer

A station in a Speaker Call connection cannot initiate a transfer.

Call forwarding

A Speaker Call -- One-way - Broadcast will not follow call forwarding.

BUSY LINE FEATURES

A Speaker Call - One-way - Broadcast cannot be overridden by another station.

STATION PICKUP FEATURES

- Speaker Call - One-way
  - A member of a Speaker Call cannot invoke a group call pick up.
  - A Speaker Call cannot be picked up.
- Speaker Call - Fixed One-way
  - A member of a Speaker Call cannot invoke a group call pick up.
  - A Speaker Call cannot be picked up.
- COM Group Speaker Call - Two-way
  - A member of a Speaker Call cannot invoke a group call pick up.
  - A Speaker Call cannot be picked up.
- COM Group Speaker Call - Fixed Two-way

  - A member of a Speaker Call cannot invoke a group call pick up.
  - A Speaker Call cannot be picked up.
- COM Group Call
  - A member of a COM Group call, can pickup a pending group call pikkup call.
  - A COM Group call can be picked up.
- Speaker Call - Two-way (Voice Calling)
  - A member of a Speaker Call cannot invoke a group call pick up.
  - A Speaker Call cannot be picked up.
- Speaker Call - One-way - Broadcast
  - A station involved in a Speaker Call - One-way - Broadcast can invoke a group call pick up after establishing a two party connection.
  - A Speaker Call - One-way Broadcast cannot be picked up.

Subscriber Hunting Arrangements

Speaker Call - One-way - Broadcast

A Speaker Call -- One-way - Broadcast will not follow redirection. (hunting)

Multiline Preference

- Speaker Call - One-way
  - Terminating Line Preference will not be checked.
  - Pre-Selection releases no Speaker Call One-way.
- COM Group Speaker Call - Two-way
  - Terminating Line Preference will not be checked.
  - Pre-Selection will not release a COM Group Speaker Call - Two-way
- Speaker Call - One-way - Broadcast
  - Terminating Line Preference will not be checked.
  - Pre-Selection (non-single button) release no Speaker Call One-way - Broadcast.
  - A Speaker Call - One-way - Broadcast cannot be bridged into until a two-way connection is established.

3.5.10 Displaying the Key Functions

The key functions of the digital system- or IP telephones can be indicated on the display with the aid of the check key.

- The key function is interrogated by pressing the check key and then dialling the code for "program key function". The "interrogate key function" mode remains active until the check key is pressed again.
- The "interrogate key function" mode is possible while a call is in progress. While it remains active, an existing call cannot be influenced by pressing the keys that can be interrogated. The display shows the text "KEY FUNCTION:". By pressing the key which is to be interrogated, the function of this key is displayed in the second display line.
- In the case of keys that can be reprogrammed from the digital system- or IP telephone, the display: "NEW CODE?" appears after 5 sec. The new key function can be determined by selecting the code and then stored by pressing the check or DUE key. Prior to storage it is possible to display another key's function by pressing it. The initiated programming procedure is ignored.
- The following keys' functions cannot be interrogated:
  - Check key (if this key is pressed the check function is terminated)
  - Loudspeaker key (local terminal function)
  - Local repertory key (local terminal function)
  - Headset key (local terminal function)

3.5.11 Programming the Key Functions

The functions of all the keys of the digital system- or IP telephones can basically be freely assigned on an individual-station basis (this refers to administrative assignment via AMOs). This includes certain keys whose functionality can be programmed or reprogrammed at the terminal itself, i.e. without the aid of the administration (e.g. repertory key, timed reminder, call forwarding key etc.).

- Keys which <u>cannot be reprogrammed</u> at the terminal can be recognised by the fact that the function display remains after they have been pressed in the "interrogate key function" mode. The "interrogate key function" mode is activated by pressing the check key.

- Keys which can be reprogrammed at the terminal can be recognised by the fact that the display "NEW CODE?" appears 5 sec. after they have been pressed in the "interrogate key function" mode. The "interrogate key function" mode is activated by pressing the check key and entering the appropriate code.
- Key functions cannot be cancelled by pressing the "clear" key. Only selective reprogramming is possible. This prevents the key functions from being cancelled accidentally.

3.5.12 Function Keys

3.5.12.1 Functions

HiPath 4000 provides a feature key for most features on digital system- or IP telephones.

English Parameter	English Definition	German Parameter	German Definition
ACC	Account Code (ACD)	ACC	Account Code (Gespr. markieren) (ACD)
ACDAGTM	Send message to agent (ACD)	ACDAGTN	Nachricht zum Agent senden (ACD)
ACDWORK	Agent work (ACD)	ACDARB	Agent arbeitet (ACD)
ACDLOG	Agent LOGON/LOGOFF (ACD)	ACDLOG	Agent LOGON/LOGOFF (ACD)
ACDEMMSG	Emergency messaage (ACD)	ACDNOTN	Notfallnachricht senden (ACD)
ACDNAV	Agent not available (ACD)	ACDNVB	Agent nicht verfügbar (ACD)
ACDPGS	Primary group status (ACD)	ACDPGS	Primärer Gruppenstatus (ACD)
ACDPQS	Primary queue status (ACD)	ACDPQS	Primärer Queue Status (ACD)
ACDSGS	Secondary group status (ACD)	ACDSGS	Sekundärer Gruppenstatus (ACD)
ACDSPVM	Send message to supervisor (ACD)	ACDSPVN	Nachricht z. Supervisor senden(ACD)
ACDSQS	Secondary queue status (ACD)	ACDSQS	Sekundärer Queue Status (ACD)
ACDAV	Agent available (ACD)	ACDVB	Agent verfügbar (ACD)
ACK	Message acknowledgement (ACD)	ACK	Nachricht quittieren (ACD)

English Parameter	English Definition	German Parameter	German Definition
DND	Do-not-disturb key	ANS	Anrufschutztaste
IUSE	Show line key I use (KEY)	ANZLEIT	Anz. d. benutzten Leitungstaste(KEY)
KNOVR	Knocking override key	ASAK	Taste für Aufschalten, Anklopfen
VCR	Voice call reject key	ASSCH	Direktansprechschutztaste
FWD	Forwarding key	AUL	Anrufumleitungstaste
PU	Call pickup	AUN	Anrufübernahme
PUS	Call pickup (executive/ secretary) key	AUNS	Anrufübernahme (bei Chef-Sekr.-App)
AUTOM	Automatic call acceptance (ACD)	AUTOM	Automat. Gesprächsannahme (ACD)
MB	Mailbox key	BK	Briefkastentaste
MSGR	Messenger key	BOTE	Botenruftaste
CH	Check (display memory contents)	CH	Check (Abfragen v. Speicherinhalten)
CL	Clear (clear display)	CL	Clear (Löschen)
VC	Voice call key	DA	Direktansprechtaste
DSS	Direct station selection key	DR	Direktruftaste
STO	Data transfer (store)	DUE	Datenübertragung (Speichern)
EXCLHOLD	Exclusive hold (KEY)	EXHALTEN	Exklusives Halten (KEY)
VACANT	Not in use	FREI	Taste nicht belegt
HOLD	Hold (KEY)	HALTEN	Halten (KEY)
CONF	Conference key	KF	Konferenztaste
LINE	Line key (KEY)	LEITUNG	Leitungstaste (KEY)
SPKR	Loudspeaker	LS	Lautsprechertaste
SPLT	Two-way splitting (flip-flop/ toggling)	MA	Makeln
MONTONE	Tone monitoring (ACD)	MITHOERT	Mithören mit Ton (ACD)
MUTE	Mute key	MUTE	Mikrophonstummschalte-taste
NAME	Name key (repertory key)	NA	Namenstaste
NV	Non-voice (data) key	NV	Non-Voice (Daten) Taste
PARK	Park key	PA	Parktaste
PRIVAT	Privacy (KEY)	PRIVAT	Privatgespräch (KEY)
CC	Code calling key	PS	Personensuchtaste
CONS	Consultation hold	RF	Rückfrage
CBK	Callback (reserving outgoing trunks)	RR	Rückruf, Vormerken ext. Leitung

English Parameter	English Definition	German Parameter	German Definition
RNGXFER	Ring transfer, executive/ secretary	RU	Rufumschaltung (Chef-Skr.-Apparat)
RCUTOFF	Ringer cutoff (KEY)	RUFAUS	Rufabschltng. d. Leitungstasten(KEY)
HT	Hunt group key	SA	Sammelanschlusstaste (Ein/Aus)
SCROL	Scroll message (ACD)	SCROL	Nachricht blättern
HS	Headset key	SG	Sprechgarnitur Taste
LANSEL	Language selection key	SPRWA	Sprachwahl
ST	Start key	ST	Starttaste
REMIND	Reminder key	TE	Termintaste
RLS	Release	TR	Trennen
DOOR	Door busy indication key	TUER	Türtaste
TRNS	Call transfer (pass over or take over)	UEG	Übergeben, Übernehmen
REP	Representative/deputy for secretary	VTR	Vertreter-Taste (Chef-Sekr.-Apparat)
LNR	Last number redial	WAWIL	Wahlwiederholung letzte gewählte RN
SNR	Stored number redial	WW	Wahl speichern, Wahl wiederholen
ADDON	Add-on conference/witness facility	ZZU	Zeugen-Zuschalte-Taste

Parameter	Definition	Usage in HiPath 4000
BUZZ_KEY	Buzz a predetermined number	JA
CONNECT_KEY	Connect to call on hold	JA
INTERCOM	Intercom feature	JA
MAIN_MENU	Main Menu key (Optiset 'E')	JA
PAS	Call Park key - System	JA
PHONEMAIL_KEY	Phonemail key	JA
PREVIEW_KEY	Preview key	JA
REPEAT_ID	Repeat party ID (ACD)	JA
SPKR_ONE_WAY	Speaker Call, Dial one-way	JA
SPKR_ONEWAY_BCAST	Speaker Call, Dial one-way broadcast	JA
STN_SPEED_KEY	(Station) speed dial number	JA
SYS_SPEED_KEY_1	System speed key #1	JA
SYS_SPEED_KEY_2	System speed key #2	JA

Table 3 New keys in HiPath 4000

Parameter	**Definition**	**Usage in HiPath 4000**
VOL_DN	Volume Down key	**Feststehende Taste,** nicht veränderbar
DCPA	Direct call park key	JA
VOL_UP	Volume Up key	**Feststehende Taste,** nicht veränderbar

Table 3 New keys in HiPath 4000

3.5.12.2 System/Station Speed Number - Chaining

It is possible to activate multiple features and functions (e.g., Call forwarding and DND) in one Station or System Speed Dial sequence. Additionally, if multiple features require more digits then are available in a single Speed Dial entry, the user is allowed to link one Station Speed entry to another.

A user is allowed to configure a Speed Dial entry to link to another Speed Dial entry. It is allowed to link up to 10 Station Speed Dial entries, in any order.

Example 1:

Speed Dial Entry 01 linked to Speed Dial Entry 11

(Note: The access code for Speed Dial access is "*55".)

Entry 01: *3 (PIN access code) + 12345678901 (PIN) + *55 + 11 (next entry)

Entry 11: 9 (LCR access code) + 15619231705 (Destination)

(Each entry 22 digits maximum)

Example 2:

PIN access code + PIN + LCR access code + Destination Number

(Each entry 22 digits maximum)

Capacities

A Maximum of 10 Speed Dial entries can be chained (each entry 22 digits maximum).

Impacted devices

Speed Dial is supported for analog telephones, digital system- and IP-telephones and Keysets. Resrictions

Restrictions

This feature is not avalable for attendants.

3.5.12.3 Time and Date Key

This feature allows a station user with a digital system- or IP-telephone with display to view the current Time and Date (for a pre-configured display time), when idle or active in a connection. The station user does this by pressing the "TIME" feature key.

Time and Date interacts with many features and call states. Time and date display is available for most call states as seen by the user.

- dle (e.g., overwrites call forward indication)
- All Talk states
- All Consultation/Toggle states
- All Hold states
- Conference states

Time and Date display format is based on the idle display format.

Restrictions

Time and Date interacts with many features and call states.

Time and Date are not available for:

- All Dial states
- Program/Service mode
- functional devices (e.g., Profiset/DSS1)
- Attendant console.

3.5.13 Door Busy Display for Digital System Telephones

A key on digital system telephones can be administered (AMO TAPRO) such that an external function (e.g. controlling an indicator with the function 'Door busy display') can be performed with it. This key is assigned an LED operated in the toggle mode.

3.5.14 Digital System Telephones without Display

The user interface for digital systemtelephones in the HiPath 4000 system is designed for digital terminals with displays. There are no audible alerting tones and only a few announcements for digital terminals. Activating, deactivating and programming a feature is acknowledged by display messages or by switching an LED on or off.

The user interface for digital systemtelephones without displays differs from the standard interface for terminals with displays in that users receive similar audible acknowledgment tones or announcements for activating, deactivating and programming a feature as analog telephone users. However, this does not mean that the user interface is the same as that of an analog telephone. The special features offered by terminals (handsfree dialling, repertory keys, etc.) are fully supported.

The following features in particular have been adapted for use on digital systemtelephones without displays:

1. Fixed and variable call forwarding: activating, deactivating and programming
2. Callback: setting and cancelling
3. Repertory keys: programming and erasing
4. Save number redial: programming, recalling and deleting destinations
5. Do-not-disturb: activating and deactivating
6. Hunting groups: activating and deactivating group allocation
7. Speed dialling: programming individual speed dialling numbers
8. Timed reminders: programming and erasing
9. COS switchover
10. Manual PIN entry
11. Function keys: reprogramming
12. Display messages: during call setup, switchover to messages used in analog telephones

    If a dialled destination is not available (except if the station is busy or the user does not answer), a digital systemtelephone with a display will output an additional advisory message with the call number and name of the dialled destination (e.g NOT AUTHORIZED, DO-NOT-DISTURB, PLEASE REDIAL etc.).
    Users with analog terminals or with digital systemtelephones without displays receive an audible alerting tone or a short announcement, e.g.:

    – This function is not possible

- The party you have called is not available
- This function is currently not implemented
- Wrong access authorisation

3.6 Special User Functions for the users

3.6.1 Display Suppression

The display suppression feature enables the user to either display or suppress the output of his data at the partner station, depending on the type of his own station (normal or unlisted).

The call number and the name of the A-user are displayed to the B-user either directly or indirectly in conjunction with the following internal traffic restrictions:

- Internal/external calls (external for ISDN station)
- Attendant console traffic
- Callback (especially from a mailbox when a party is free)
- Call pickup (camp-on, call pickup group)
- Call forwarding - all calls/no answer
- Server traffic
- Connection to paging system

The following distinctions are made, depending on the type of subscriber line circuit:

1. Normal subscriber line circuit with occasional display suppression.
    - This is the standard type of circuit. When calling in ISDN, the call number and name of the A-subscriber are displayed on the connection partner's terminal.
    - The display of the call number and the name of the A-user at the partner station can be suppressed on a case-to-case basis.
2. Unlisted station with option of occasional display.
    - Intended for users who wish for permanent secrecy. The call number and the name are not indicated to the called partner.
    - Suppression of the call-number and name display at the station of the called user can be cancelled on a case-to-case basis.

3. If the display is suppressed, the terminal of the called user receives the following signals:
   - No text
   - No "blank signal"
   - Display of 5-position symbolic information: "*XXXXX*"
4. The display suppression on/off function is active for the following equipment types:
   - analog telephone
   - digital system telephone
   - IP telephone
   - attendant Console
5. Network-wide function only available via CorNet.NQ protocoll.
6. Extension of the user interface:
   - Display suppression off" code number for unlisted stations
   - Display suppression on" code number for normal stations
   - No separate feature keys
   - Code number can be stored with the name key
   - Additional display for the calling user (not analog telephones): 'NO NUMBER DISPLAY/NUMBER DISPLAY ON' + dial tone as a positive acknowledgment. The display function can only be switched at the start of a call, i.e. the display-suppression code number can be dialled:
     – When a connection is set up in the idle state (digital system- or IP telephone) or after going off-hook in the "dial tone" state. The dial tone is repeated when the code number is dialled.
     – When a connection is released with the release key on the digital system- or IP telephone.
7. If the display function is switched over, the new setting is only valid for the duration of the call. If anonymity or cancellation of anonymity is desired frequently, the other operating mode should be selected instead.
8. Display suppression is ignored in conjunction with the following features:
   - Calls to the paging system (display)
   - Call tracing: The A-call number and the name are always included in the call tracing report.

9. When a callback is delivered (free/busy), the display function which is valid is not that which was set for the original call, but the default setting for the A-user (= the station which is called back).
10. Callback when a party is free with an entry in the mailbox of the B-user is only possible for a normal station with display suppression switched off (default setting). Users who are administrated as "unlisted stations" are not able to employ this feature.
11. Feature functions which are not possible in conjunction with display suppression are rejected. These include:
    - Paging via an unlisted station (A-user) if an undesired connection is set up to a paging system and the B-user has activated call forwarding (all calls) to this system. The caller hears a busy tone, without being able to initiate features.
12. CID-B/U for analog or T1 trunks

    Automatic number identification (ANI) is a public network service which is used in the CO serving the originator of the call when the presentation indication of the caller is not delivered over the trunk.
    Via the CID-B/U feature it is possible to control if an ANI-Number has to be generated in the CO or not.

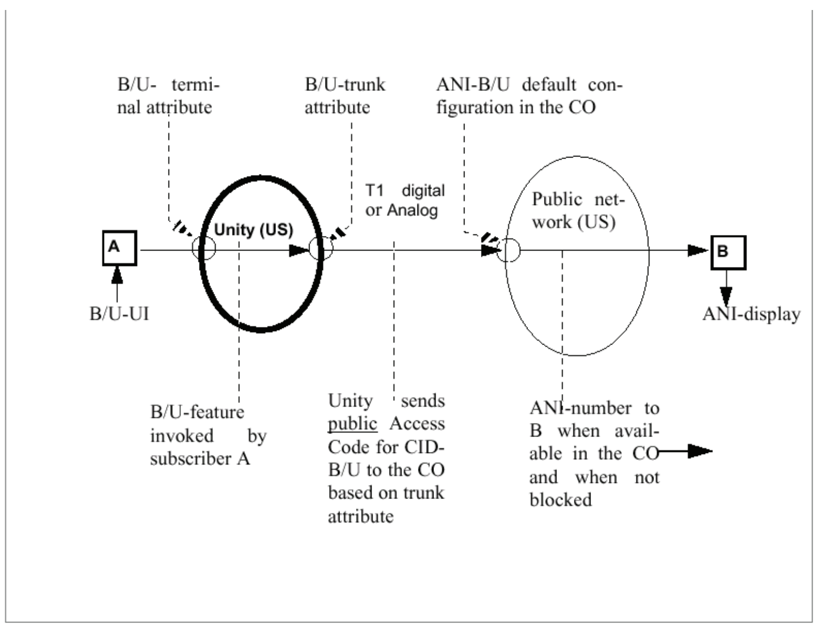


Figure 22 B/U of ANI number in the CO

Calling Party:

B/U-Terminal Attribute: Describes if the Subscriber has blocking or unblocking CID per default.
B/U-UI: Override of the default setting on a call with the Feature Code

Outgoing trunk:

It depends on the following parameters which codes has to send to the CO:

- If HiPath 4000 has to send the access code for CID-B/U depends on the default configuration of the trunk in the CO which can be ANI-number is blocked (per default) or ANI-Number is unblocked (per default)
- Which code depends on the trunk signaling type which can be DP (Dial Pulse) or DTMF

3.6.2 Mailbox

Users with digital system- or IP telephones can call up requests for "callback on no answer" stored for their station before these requests are carried out, and then carry them out selectively.

Whether or not station B has the "mailbox" feature does not influence entry of the callback requests on no answer.

If a station user has the "mailbox" feature, the time of the callback request entry is additionally stored with the callback request for his station.

The following applies as a "callback on no answer" is no longer entered in the mailbox for terminals without a display. This restriction is required because users with terminals without displays cannot distinguish whether the indicated entry is a voice mail message or a callback.

A maximum of 100 % of station users can be authorised for the "mailbox" feature. The only requirement is that they have a mailbox key.

The "mailbox" feature can be used not only to scan the list for entered requests for callback on no answer for that station, but also to "scan" the lists of entries in the Infobox for voice, and fax.

The LED for the mailbox key is lit and the announcement "message waiting" is given each time an outgoing seizure is made as long as there is at least one entry in the mailbox for any of the services which has not yet been carried out (callback not yet carried out, message not yet answered or delivered).

Interrogation of entries in the mailbox.

- – The contents of the mailbox can be shown on the display by means of "scanning", i.e. by successively pressing the mailbox key.
- – The mailbox entries can be interrogated with or without user identification.
- – The memory contents are displayed cyclically, i.e. after the last entry has been shown, the display begins again at the start.

Carrying out a callback following scanning in the mailbox

– By pressing the ST key the station user can establish a connection to the station whose call number is shown on the display during scanning without having to dial the call number.

– Setup of an outgoing connection in the programming mode by means of pressing the ST key does not terminate this state.
After the connection which has been set up by pressing the ST key is ended, the next entry in the mailbox is automatically shown as long as the programming state was not terminated with the CH key or the mailbox key used to scan further.

– The callback to station A which is activated during scanning in the mailbox by station B ignores the features do-not-disturb, call forwarding, call forwarding to the secretary in an integrated executive/secretary system, call forwarding - no answer and call forwarding - no answer in hunting group.

Deleting a callback request which is entered for a station with the "mailbox" feature can be carried out by

– manual deletion by station A,

– entry of a new callback request by station A (analog telephone),

– pressing the ST key by station B to carry out the callback,

– manual deletion by station B during scanning.

3.6.3 Trunk Call for Extended Calls

HiPath 4000 can be administered (AMO ZAND) in such a way that a trunk call extended from the attendant console to an extension or a consultation call from a trunk call is signalled at the called user's station as a trunk call or internal call.

3.7 CLIP for analog telephones

Display of telephone number and name

Analogue telephones can indicate telephone number and name of calling subscriber. The feature contains the display of telephone number and name of calling subscriber (as far as available). The CLIP information is distributed to the device between first and second ring tone. According to the calling type telephone number and name of calling subscriber are shown or just the telephone number.

With an external Call from a public or private network-interface , a fixed name can be indicated.

The CLIP- Information is distributed allways to the called subscriber and only once a time.

Restrictions

- The Feature Clip for analogue telephones can only be activated for the whole telecommunication system. With it the SLMAE-boards are provided with the CLIP information in calling status. The CLIP Information is distributed from SLMAE to all devices, independent whether there is an CLIP able telephone or not. All Ports of the board are treated equal.
- There is neither a parallel display nor an update of the display. That means within toggling and call transfer loosing the Information from the calling subscriber.
- The Feature CLIP for analogue telephones can be provided with a combination of the features Call Forwarding, Hunt Group, Pickup Group, Conference Call, Consultation Call and Hotline.
- The ETSI Standard with FSK Modulation respectively DTMF Signalling and Bellcore Standard with FSK Modulation is supported.

VMWI for Clip for analogue user

This feature is provided in combination with call back and/or mailbox. The CLIP Information to the called subscriber contains the Telefone number of the calling user and the demand VMWI (Visual Message Waiting Indication) -> activate LED.

Call Back and Mailbox enquiry takes place according to the configuration with the corresponding Featurecodes. Die VMWI LED is turned off with the corresponding CLIP information being no call back request or mailbox-message any more..

The feature is activated after installing SLMAE and activating CLIP function for analogue telephones.

Restrictions

The MWI/Mailbox Signalling with glow lamp (150 V) is not supported.

4 System Features

4.1 DTMF Transmission to the Public Exchange

4.1.1 DTMF Transmission to the Public Exchange on digital Connections

Outgoing external two-party calls can be set up via administration and maintenance such that signals are converted into DTMF control information for necessary suffix dialling. This feature permits the control of subsequently connected special equipment, such as City-Call, voice mail systems, dictation equipment etc., from a digital system- or IP telephone.

1. Conversion takes place for the following connections:
   - CO trunks (all types) and
   - tie trunks (all types) with
   - analog transmission or
   - digital transmission (S_0/S_2, DSS1/CorNet NQ).
2. Conversion takes place in the case of analog and digital lines using 'in-flight injection' of the DTMF signals.
3. The following configurations can be set up via administration and maintenance:
   - Routing characteristics for outgoing connections
     – services possible using suffix dialling or
     – transparent (unchecked) digit forwarding to outside.
   - Operating mode per route
     – after trunk seizure through DTMF transmission without request procedure or
     – DTMF transmission through suffix dialling procedure.
   - Display per route (after simulated or real end-of-dialling)
     – display of the digital signals converted into DTMF signals or
     – suppression of each display or
     – standard display (e.g. DTMF transmission).

4.1.2 DTMF Signalling to Public Network on Analog Connections

Special services in the public network exchange such as City-Call, VMS systems, dictation equipment etc. can be controlled with analog voice terminals within the HiPath 4000 system. For this, the dialled signals for suffix dialling are converted from pulse dialling signals to DTMF signals.

This feature is activated by either

– dialling a code (call-up procedure) in talking state or

– by selectively seizing routes which lead via DTMF converters for suffix dialling conversion.

The call-up procedure can be administered via AMO. Suffix-dialled DP signals (0 - 9, *, #) are converted to DTMF signals before being outpulsed to the exchange.

The feature can be used in the following call states:

– outgoing external two-party connections

– external connections which have already been switched through

– transit connections

The feature applies to exchange (central office) trunks and tie-trunks with

– analog signalling or

– digital signalling (for DSS1 / CorNet NQ / SWISSNET2 protocols via S_0/ S_2 interfaces)

4.1.3 Dialling

User features are activated by

– dialling a code,

– actuating feature keys.

4.1.3.1 Digital Pushbutton Dialling

1. Digital transmission method for dialling information on the station line circuit

   - for digital systemtelephones,
   - IP telephones,
   - for the attendant terminal.

2. by 12-element pushbutton set
3. A device code is allocated for each line with the aid of administration and maintenance.
4. The predetermined keys of the digital system- or IP telephones are uniformly defined with respect to their meanings. It is possible, however, to assign any (useful) function contained in the overall scope also to the predetermined keys.
5. It is not necessary to press any key (equivalent to the signal key for dual-tone multifrequency signalling; possible for all services except callback with single digit dialling) to initiate suffix dialling with the aid of the pushbutton set or a feature key.
6. Suffix dialling during the ringback tone or busy tone (to activate a feature)
   - After the last digit has been dialled a positive or negative acknowledgment is shown on the display (for a maximum of 5 s).
   - If the called user answers during dialling, any further digits will be ignored (i.e. the selected feature is not activated).
7. Changeover of the pushbutton set of a voice connection from a digital systemtelephone to a non-voice connection or to the programming mode:
   - Changeover takes place by pressing the non-voice key.
   - The non-voice lamp lights during the changeover state
   - Speech-accompanying setup of the non-voice connection or programming mode.
8. Changeover of the pushbutton set of a non-voice connection from a digital systemtelephone to a voice connection:
   - Changeover takes place by going off-hook (outgoing seizure) or, in a speech- accompanying non-voice connection, by pressing the QUERY, REJECTION or TRANSFER key with subsequent dialling tone to another voice connection.
   - The non-voice connection is automatically placed on hold when changeover takes place.
9. Changeover of the pushbutton set on digital systemtelephones from the programming mode to a voice connection (digital system- or IP telephone):
   - Changeover takes place by going off-hook (outgoing seizure) or, in speech-accompanying programming, by pressing the QUERY, REJECTION or TRANSFER key with subsequent dialling tone to another voice connection.

- In case of digital systemtelephones, the programming mode is automatically terminated when changeover of the pushbutton set takes place.

10. Anon-voice connection or the programming mode can be active, not both together. The keystroke to initiate the other function is ignored if an attempt is made to have both functions active at the same time.

4.1.3.2 Dual-Tone Multi-Frequency (DTMF) Signalling

- Dual-tone multifrequency signalling as per ITU Q.23 on the station line.
- by 12-element pushbutton set
- by 12-element pushbutton set with 4 additional keys (feature keys). The four additional keys send frequency combinations 13 to 16 with or without additionally generated flash. They are interpreted in the system as appropriate features for each individual user.
- 100 % of the stations can be equipped with DTMF telephones.
- The signal key is pressed to initiate suffix dialling:
- Each station line is capable of interpreting the ground key and flash key as suffix dialling criteria.
- In certain cases a signal key is not necessary in order to initiate suffix dialling. The code receiver remains ready to receive for a certain period of time after the end of dialling.
- To initiate suffix dialling on the comfoset 160 telephone it is not necessary to press a signal key if the four additional keys of the 12-part dialling key block are being used since these feature keys have an integrated flash facility.
- In cases where dial tone, busy tone or ringing tone are sent after a negative acknowledgment another procedure can be selected during or after the negative acknowledgment.
- The system is designed such that a loop interruption of $\geq 40 < 200$ ms can be evaluated as flash. These times can be administrated, so that any regulations can be employed system-individually within the specified framework.
- When the connection has been set up the system is transparent for all DTMF frequencies.
- If central equipment (such as the code receiver) is temporarily unavailable for initial seizure or suffix dialling, the user can wait off-hook until he hears the internal dial tone (waiting time system).
- A flash keystroke or an integrated flash is ignored while a code receiver is connected.

4.1.3.3 Rotary Dialling

1. Dial pulsing on the station line
   - for rotary dial telephones
   - for dial pulsing pushbutton telephones
2. 100 % of the stations can be equipped with rotary dial telephones and dial pulsing pushbutton telephones.
3. Automatic DP or DTMF signal detection is possible.
4. The signal key is actuated to initiate consultation. Ground potential is thus connected to the system (3-wire connection).
5. Consultation hold on terminals without a signal key can be initiated by dialling a one-digit code (2-wire connection). The same code can then be used to cancel consultation hold, transfer and hold toggle. This code has no relationship to other codes.

   If the signal key is replaced by the dialling of a code, the following restrictions apply to cancelling consultation hold:
   – Digit input: Cancellation only possible as long as dial tone is heard.
   – External dialling: Cancellation only possible after end of dialling.
   – Speed dialling facility: Cancellation only possible after end of dialling, except for internal destination.
   – PSE: Cancellation only possible after end of dialling.

   In all cases specified it is possible, however, to return to the waiting subscriber by going on-hook with immediate recall.
6. It is not necessary to press a grounding key to initiate suffix dialling (applies to all services, but not to consultation).
7. Suffix dialling during the ringback tone (to activate a feature)
   - Ringback and ringing tone are terminated when the first digit is dialled.
   - After the last digit has been dialled the calling user receives a positive or negative acknowledgment (acknowledgment tone or recorded announcement).
   - In cases where dial tone, busy tone or ringing tone are sent after a negative acknowledgment, another procedure can be selected during or after the negative acknowledgment.

4.2 System Operation with/without direct inward calling

4.2.1 Direct Inward Dialling

- Direct inward dialling circuits are directed.
  - Incoming or
  - Incoming/outgoing
- Direct inward dialling is possible to:
  - Internal stations
  - Attendant consoles
  - Hunting groups
  - Voice Mail
  - Facsimile users
  - Data terminal equipment
- Call forwarding on direct inward dialling calls

  Depending on communication type and the status of the dialled destination, unsuccessful DID calls are diverted to a specific permanently connected destination.
  - Alternatively, music or an announcement can be applied to the waiting user.
  - Waiting users can be connected via TMOM to answering devices for synchronised announcements.
- If call forwarding on direct inward dialling calls is not activated, the external user receives audible tones.

  Station terminals which are to be obtainable by direct inward dialling are given DID authorisation:
- Up to 100% of users can have DID authorisation.
- DID authorisation can be temporarily prevented
- In incoming trunk calls the attendant console is obtained by dialling the DID number and then the attendant console code.
- Stations are obtained by dialling the DID number and then the station number.
- Certain permanently connected stations are obtained in incoming trunk calls with single digit station numbers.

4.2.2 Inward Restriction

Stations with trunk access can be excluded from direct inward dialling; they can still be obtained with extended trunk calls.

- Inward restriction is carried out with the administration and maintenance system by entering inward restriction for the user in question.
- Inward restriction can also be carried out by class-of-service changeover (from the attendant console or automatically).

4.2.3 System Operation without direct inward dialling

If the system is to be operated without DID, the DID equipment in the system is deactivated.

1. In incoming trunk calls the attendant console is obtained in the same way as in systems in which DID has not been deactivated, i.e. with the digit 0 with two-digit numbers from 01 to 00 or any other user-defined number.
2. In systems in which DID has been deactivated, certain permanently connected stations can be obtained with single-digit station numbers from 1 to 9 as in DID.
3. Operation without DID is implemented by the system diverting all DID calls, irrespective of what digits are dialled after the DID number, to the attendant console, with the exception of single-digit station numbers.
4. Dedicated incoming trunks
   If exchange circuits without DID are used, the "dedicated incoming trunks" feature can be used to signal calls on certain CO trunks at certain stations, bypassing the attendant console.

   CO trunk and station are assigned to each other with the administration and maintenance system.

   – A maximum of all CO trunks can be set up for dedicated incoming trunks.
   – All CO trunks can be assigned to a station.
   – A CO trunk can only be assigned to one call number.

   The following destinations can be assigned to a trunk:

   – Station
   – Hunting group

   Prevention of incoming dedicated trunks

- Called station is free, user does not answer: ring tone immediately and forwarding after an interval.
- Called station is busy, not connected, unallowed call, do not disturb: diversion and ring tone immediately.
- Calls are diverted to the attendant console; at night: signalling in accordance with night service variant.
- In diversion after an interval, the forwarding destination and the station are called in parallel. If the forwarding station or the called station answers, ringing is ended at both station terminals.

Dedicated incoming calls take call forwarding into account. They can also be interrogated by call pickup at another station.

5. Announcements with suffix-dialling

   For some situations in which the required destination cannot be reached by direct dialling, callers can be connected to a synchronous announcement with subsequent DTMF suffix-dialling.

4.2.4 Partial Inward Restriction

In systems in which DID has been deactivated, certain permanently connected stations can be obtained as in DID with single-digit station numbers from 1 to 9.

The administration and maintenance system is used to assign the single-digit station numbers from 1 to 9 to the station or automatic line testing circuit to be reached with these numbers as in DID.

If an external caller dials a number not assigned to a station he is diverted to the attendant console.

4.3 Technical Provisions for the Prevention of Calls

4.3.1 Prevention of Undesired Calls

Calls can be prevented depending on origin and destination.

1. The administration and maintenance system can be used to define what connections between groups of stations in the PABX are to be permanently prevented.

   Internal traffic restriction (ITR) groups can be formed consisting of a combination of station groups.

   Groups taken into account when the prevention matrix is set up:

   - User groups:
     Tenants of a shared system
     Station groups or user classes (closed user group) for non-voice services
   - Attendant groups
   - Standby attendant positions/night attendant consoles
   - Night stations (night service variants activated simultaneously)
   - Trunk groups

   User classes-of-service and trunk types (incoming or outgoing) are not taken into account when setting up the prevention matrix.

2. The ITR group derivable from the PIN and a special matrix can be used to define whether an ID card may be used in another user group. This matrix is independent of the ITR matrix.
3. Connections can be prevented with the following means other than the prevention matrix:
   - Classes-of-service (e.g non-exchange access authorisation prevents trunk calls)
   - One-way trunks (e.g. an outgoing trunk prevents incoming traffic)
   - Temporary prevention by class-of-service changeover
   - Digit analysis groups: individual code assignment

4.3.2 Preventing Invalid Connections

In networked systems with CorNet NQ signalling, the access code of the source node is transmitted with the calling line identification message in each call setup request, and checked against the destination node access code in each transit node via the number matrix. If the number matrix is configured accordingly, this can prevent invalid connections from being set up.

The source node access code is checked against the destination number node in each transit node as the call setup request is received.
The connection setup types checked in this manner are basic call setup, consultation call setup and call transfer (transfer from consultation hold and attendant connect).

The check can be cancelled for attendant connect transfers by setting the feature bit string via AMO.

Incoming calls from the public network exchange are assigned a virtual source node number (if several nodes have exchange access, several virtual codes will exist). This virtual node access code is then checked against the destination node access code in the transit nodes for the same purpose.

Connections from other nodes or networked systems without CorNet NQ signalling are also assigned a virtual node access code which identifies the incoming trunk group.

4.4 Shared systems

A system can be shared between several customers (independent companies). Connections between the user groups (tenants) are normally only possible via the public network. The system equipment (trunk groups, attendant consoles etc.) can be shared or used separately by each tenant.

The number repertoire is available for shared use by all user groups.

Each user group has its own range of numbers.

Optionally joint attendant console or attendant group for each tenant.

Common attendant console for all tenants.

- – Attendant console display during call signalling indicates the tenant for which a call is intended.
- – External calls are extended to all users in the system.
- – Attendant calls are signalled at the common attendant console. The attendant code is the same for all users in the system.
- – The attendant can set up calls not produced during extending to any user in the system.

Attendant console group for each tenant

- – Normal call signalling when calls are made via exchange circuit and tie-line circuit.
- – Attendant calls are signalled at the associated attendant group. The attendant code can be the same for all users in the system.

- Connections can be set up independently of the ITR matrix from any attendant group to users in any user group. During call extending the resulting call (from the caller to the station) is then checked for permissibility in accordance with the ITR matrix.
- Attendants can use all trunk groups in the system for outgoing seizure of external trunks independently of the ITR matrix.

- A shared system has full flexibility of assignment of attendant consoles or night stations to tenants (the following configuration is also possible: attendant console for each tenant during the day, common night station for several or all tenants during the night).
- Attendant groups can be selectively obtained by external users with codes 001 to 000 or on the basis of the trunk group (this applies to CO and tie trunk calls).
- The tenant name is shown on the attendant console display even if the attendant group only receives calls for one tenant.
- Where the attendant group responsible for an incoming external call cannot be ascertained by the system, e.g. if dialling is incomplete, the call is diverted to any attendant group defined for this case with the administration and maintenance system after a timeout.
- For calls via tie-line circuit, after the standard attendant code has been dialled the common attendant console or the appropriate attendant group is obtained depending on the attendant console configuration.
  - The administration and maintenance system is used to define from which tie line or station line groups an attendant group is to be obtained.
- There is a maximum of one night station per attendant group.
- Each attendant group has a maximum of 8 separate night service variants.
- A common attendant console only has common night service variants for all tenants in a shared system.
- Certain tenants can dispense with an attendant console for their user group. On system startup the first night service variant of this tenant is activated (permanent night service). A fictitious attendant group is assigned to this tenant.
  The night service for this tenant is provided by activating call forwarding - all calls at the attendant console; one or more attendant groups.
- Different console answering priorities are possible for each tenant.
- Class-of-service changeover is possible from every attendant console for each class-of-service changeover group (the shared system does not affect this feature).

  Call data registration
  - CDRA is possible for each tenant
  - CDRS is common for all tenants
  - Immediate printout on common terminal
  - Common call charge computer for all tenants

- The central system speed calling directories can be obtained by one, more than one, or all tenants, as in a normal system. If there is a trunk group for each tenant, a trunk in the own trunk group is seized.
- Any assignment of trunk groups to tenants is possible. There can be an outgoing CO trunk group for each tenant.
- Tenant groups can use the same trunk group codes, which means that users automatically obtain a trunk group assigned to their tenant group.
- Blocking combinations can be assigned to the tenant groups for toll/code restriction.
- One digit analysis list per tenant.
- Common numbering for all tenants (stations are numbered in consecutive order). Numbering can be matched to the division of stations into tenant groups with a different first digit for each tenant.
- Service code numbers can be different for each tenant (digit analysis list per group).
- Diversion of DID trunk calls to shared attendant console or to the tenant's attendant group.
- Controlled station-to-station restriction is possible from all attendant consoles for all user groups (the shared system does not affect this feature).
- Controlled station restriction - group can be activated from all attendant consoles for each of the user groups (the shared system does not affect this feature).
- The public address areas can be distributed among the tenants as required.
- Entrance telephones can be assigned to tenants as required.
- Dictation equipment can be assigned to tenants as required. Codes are assigned to these by user group-specific evaluation.
- Shared or tenant-specific paging system.
- Shared night watchman service.
- Station terminals which can be obtained by users of all tenants or from which connections to all users can be set up are combined in a separate group.
- Short recorded announcements are shared by all tenants
- Call tracing printout at the shared service terminal.
- The different ringing tones are the same for all tenants.
- Remote maintenance and administration is the same for all tenants.

- Trunk reservation is the same for all tenants (after pressing the RL key on any attendant console the trunks defined for the entire system are reserved).
- System-specific data e.g. transfer system and activatable features are the same for all tenants of a shared system.
- The administration and maintenance system can be used to enter a company code up to 4 characters long for personal identification with ID card for each ITR group.
  This prevents non-company personnel making telephone calls at the company's cost if they happen to have a PIN allowed in this system on their ID card.
- In shared systems the assignment of ringing tones to types of call is the same for all customers.

4.5 Route Discrimination

1. External lines can be divided into several trunk groups of any size.
2. Each trunk group is seized in an outgoing direction with its own code.
3. A trunk group can be reached under more than one code (with LCR feature)..
4. Trunk groups are defined by means of the administration and maintenance system.
5. The codes are allocated to trunk groups using the administration and maintenance system. Examples:
   - type of traffic
   - consultation possible
   - type of dialling (pulse dialling or DTMF)
   - directional trunk groups
     – incoming
     – outgoing
     – incoming/outgoing
   - with toll/code restrictions
   - with call charge registration
6. Seizure of a trunk group
   - from digital system- or IP telephones
     – by dialling the group access code,

   - using the trunk group key (= destination key/name key with stored group access code),
   - with last number redial or an abbreviated code.
     The group access code is stored with the external call number.
 - from the attendant console
   - by dialling the group access code,
   - using the trunk group key (= destination key with stored group access code).
     There can be up to 40 trunk group keys in the upper section of the attendant console keypad.
     There can be up to 2 remote trunk group keys in the attendant console (R1, R2).
     Depending on the entered dialling information, each of the above-mentioned keys can be used as a trunk group key or destination key.
   - with last number redial or an abbreviated code
     The group access code is stored with the external call number.
 - from an analog telephone by dialling the group access code or selecting last number redial or the abbreviated code.
 - from non-voice terminal equipment:
   - by dialling the group access code
   - with speed dialling: the group access code is stored with the external call number.

7. When an user/attendant with a digital system- or IP telephone in the idle state presses a feature key (last number redial key, repertory key, trunk group key, destination key or DSS key) without stored information, a line is seized without dialling.
   - Digital system- or IP telephones
     - User/attendant receives internal dial tone.
     - A message is shown on the display to indicate that nothing is stored.
     - The user/attendant can then dial using the keypad or other feature keys.
   - Attendant console
     - The attendant receives internal dial tone.
     - A message is shown on the display to indicate that nothing is stored.
     - The P key is automatically seized.
     - The attendant can then dial using the keypad or other feature keys.

8. At the user's option, the administration and maintenance system can be used to make seizure of a trunk group dependent on the first or first and second dialled digit of the external call number. To do this, a group access code shared by two or more trunk groups is dialled. After the system has selected the appropriate trunk group, a line is seized and the buffered digit (max. 2) is transmitted with the digits that follow. The administration and maintenance system is used to set the following:
   - check of the first and second digit of the external call number,
   - assignment of the trunk group to be seized to the digits.

4.6 Authorizations

Authorizations

1. Trunk access
   - No trunk access

     The user cannot seize either incoming or outgoing trunk lines.
   - Outward-restricted trunk access

     The user can receive incoming trunk calls but outgoing trunk calls are routed via a terminal with trunk access (attendant console, night station).
   - Inward trunk access

   Examples of inward trunk access include local access, local area access, continental access and intercontinental access.
     - The user can receive trunk calls, but outgoing trunk calls are subject to toll restriction.
     - Every toll restriction call number group is assigned a class of service.
     - Up to 6 restricted trunk access authorisations.
     - More than one restricted trunk access authorisation can be assigned for each user.
     - Restricted trunk access authorisations are, for example:
       Local access,
       Local area access,
       Continental access,
       Intercontinental access.
   - Unlimited trunk access
     No restrictions apply to the user for either incoming or outgoing trunk calls (no toll restriction).
2. Tie trunk access

- Tie trunk access with LCR
  - The user can receive incoming tie trunk calls, but outgoing tie trunk calls are subject to LCOS.
  - Up to 6 tie trunk classes of service.
  - More than one tie trunk access authorisation can be assigned for each user.
  - Every call number group is assigned a call restriction authorisation.
  - If more than one tie trunk group has been defined, the user's tie trunk access applies to all tie trunk groups to which he/she has access. For each trunk group, another call number group can be assigned to tie trunk access (via LCR).
- Unlimited tie trunk access

  The user is not subject to tie trunk restrictions either for incoming or outgoing tie trunk calls.

4.7 Discriminating Ringing

With the administration and maintenance it is possible to define whether the ringing tone to the stations for unanswered trunk calls (trunk ringing) is to be same as for other calls.

1. If discriminating ringing is not required the stations are sent the internal ringing tone in all cases.
2. Discriminating ringing is received by
   - stations in direct inward dialling (analog telephones and digital system- and IP telephones),
   - night station (analog telephones and digital system- and IP telephones),
   - transfer station for call forwarding - no answer for the attendant console (analog telephones and digital system- and IP telephones).
3. Trunk ringing
   - for DID trunk calls to the stations,
   - for trunk calls at the attendant console,
   - for night service trunk calls,
   - for forwarded trunk calls in the case of call forwarding - no answer for the attendant console,
   - for diverted DID trunk calls which have not been answered at the station:

   - calls diverted to the attendant console,
   - calls diverted to the night station.

4. Internal ringing
   - for extending,
   - for consultation hold,
   - for internal calls,
   - for tie trunk calls,
   - for callback.
5. The attendant terminal only ever receives continuous ringing
6. In addition to trunk ringing and internal ringing, other ringing tones can be assigned to certain types of call.
7. Fixed discriminating ringing
   - Assignment of ringing tones to types of call is fixed.
   - Emergency ringing
     - for calls in the timed hot-line service,
     - cadence of emergency ringing: continuous ringing (analog telephones and digital system- and IP telephones).
   - Alerting tone
     - at the start of visual call waiting indication
     - for override,
     - for camp-on,
     - for release for camp-on,
     - for timed reminders,
     - for call pickup groups,
     - for CHESE.
8. Variable discriminating ringing

   Assignment of ringing to the various call types is established as standard, but at the customer's request, call types can be flexibly assigned with the aid of administration and maintenance to special ringing 1, special ringing 2, special ringing 3 (for all users of analog telephones or digital system- or IP telephones), or internal ringing.
   - Special ringing 1 (standard)

  - for calls which arrive at a station after call forwarding - no answer;
  - hunting group calls,
  - for timed reminders,
- Special ringing 2 (standard)
  - for direct calls to analog telephones or digital system- or IP telephones,
  - for messenger calls,
  - for urgent calls at the attendant terminal (timed hot-line service calls).
- Special ringing 3 (standard)
  - depending on the calling user for standard internal connections and internal consultation
  - depending on the calling user for data controlled connections

9. In shared systems the assignment of ringing tones to types of call is the same for all customers.
10. A trunk call can be assigned to certain trunk groups with the aid of administration and maintenance (with the exception of CO trunk groups and some or all tie-trunk groups).

4.8 Attendant Intercept on DID Calls

Depending on the communication type and the status of the dialled station, unsuccessful DID calls are redirected to a specific permanently connected destination.

1. If call forwarding on DID calls is not activated, the external user receives audible tones.
2. The parameters for call forwarding are set with the administration and maintenance system.
   - Trunk group/type of traffic: trunk calls, tie trunk access -priority tie trunk, tie trunk access- non-priority tie trunks.
   - Status of dialled station terminal: see call diversions
   - Diversion destination: attendant console
   - Time of call forwarding:
     The time of call forwarding depends only on the status. It is defined for each status (immediate or after an interval).
3. Call forwarding is to the attendant console. The calls are distributed

4. During night service calls are forwarded to the night station
5. Call diversions
   - Called station is free: ring tone immediately and call forwarding after an interval
   - Called station is busy: call forwarding and ring tone immediately
   - All trunks busy (connection element temporarily taken out of service by administration and maintenance or dependability system): call forwarding and ring tone immediately.
   - Connection element not connected: call forwarding and ring tone immediately
   - Unequipped route, subscriber line circuit not available, call number does not exist: call forwarding and ring tone immediately.
   - Unallowed connection, station not authorised: call forwarding and ring tone immediately
   - Called station is barred to DID: call forwarding and ring tone immediately.
   - Called station has do not disturb; call forwarding and ring tone immediately.
   - Station number not dialled: call forwarding and ring tone after an interval.
   - Station number incompletely dialled: call forwarding and ring tone after an interval.
6. Call forwarding to an attendant console is also possible on DID tie trunks.
7. After a DID call has been forwarded, if the station is free and does not answer within a certain time the attendant console and station (including a station in the satellite PABX) are rung in parallel. If the attendant or called user answers, ringing is ended at both stations. If the attendant answers, the connection to the station is cleared down.

   If the called user is to be called again, i.e. has become free after the attendant has answered the recall, if parallel ringing is set, attendants simply press the S key in order to connect the caller in this case. The administration and maintenance system allows the option of defining whether the ringing is to continue at the called station while the recall is being answered by the attendant console (parallel ringing) or not. To do this, the attendant must press the clear key and dial the new station number, or simply dial the new number immediately (implied clearing). If parallel ringing is set, the answering is indicated to both parties as follows:

   – the attendant console receives an appropriate display message

   – the answering party receives a hold notification (announcement or music).

This feature does not apply for recalls if called user is busy or has become free and parallel ringing is not set.

8. If the number is incomplete or no number is dialled, after a timeout the call is forwarded to any attendant group defined for this purpose.
9. If in operation without DID an external user dials a station number he is immediately forwarded to the attendant console and obtains the ring tone
10. If the feature "single-digit station number in incoming central office traffic" is used, an external caller is forwarded to the attendant console if no station is assigned to the dialled number (1--).

4.9 Digit Prefixing

4.9.1 Digit prefixing in outgoing traffic

For dialling to another exchange it is possible to define a series of prefixed digits (max. 18 digits) for each trunk group.

The prefixed digits are freely selectable, and may also be a repetition of the route code. Digit prefixing can also be set for each overflow trunk group within a routing group (in addition to "separate" prefixing for the relevant trunk group).

4.9.2 Digit prefixing in incoming traffic

For an incoming connection it is possible to define a series of prefixed digits for each trunk (max. 6 digits).

The preset digit sequence is freely selectable and may also be a repetition of the route code.

4.10 User Numbering in the Network

4.10.1 Open numbering

With open numbering the individual systems (nodes) are numbered independently of each other. Each node has its own number repertoire.

A number is therefore only unique within one system. If a connection is to be set up between two nodes the number must be prefixed by a route code.

If a telephone number is dialled which belongs to a station within the user's system, the route code which belongs to this system can also be dialled along with the telephone number. When analyzing the numbers, the system recognises the fact that the destination dialled is located in the system and ignores the route code.

4.10.2 Closed numbering

With closed numbering the numbers are issued uniquely throughout the network. In this case the network behaves like a large system.

Closed fixed numbering

With this method the routing codes are part of the number and are thus transparent to the user. The number to be dialled is the same for internal and external users; it is independent of the location of the calling party in the network.

Closed flexible numbering

The destination node and the route code to this node are concealed in the user number.

The user is unable to tell from the number whether he is in a network of systems or an individual system.

The numbers must be unique throughout the network.

This means that the entire network numbering plan has to be set up in each individual node in the network.

4.10.3 Mixed open/closed numbering

It is possible to set up open and closed numbering on a mixed basis.

4.10.4 Extended numbering for networks

A network-wide, uniform numbering scheme for routes and nodes can help reduce costs, provided the following conditions are met:

- Automatic route selection via LCR. The LCR system of the switch decides on the route to be followed for connection setup on the basis of route availability and cost factors (e.g. via private network with CO breakout)
- Differentiated, individual restrictions for route selection / destination selection by users
- Uniform, open numbering scheme, i.e. user numbers are independent of their location in the network.

- Network-wide signalling of calling and called line information (A-user and B-user)
  - Modification of signalled number information
  - Modification of line number information for outbound and inbound connections
  - Modification of signalled number information depending on the home node of the caller
- Open numbering of servers and services throughout the network, in order to identify service users .

4.10.5 virtual numbering plan

The Virtual Numbering - VNR feature in HiPath 4000 supports

- the multiple assignment of station numbers,
- the configuration of overlapping station numbers with different lengths (depending on DPLN groups).

Multiple virtual nodes must be created for this in the HiPath 4000 (physical node).

The station ports to be configured are each assigned to one of the virtual nodes. The station number of a station port is now only unique within the virtual node.

The unique identification of a station within a physical node is only possible with the combination node code and station number.

Modified Display for E.164

The following display format options are available for virtual station numbers:

- Node code + station number,
- Private numbering plan + station number
- E.164 data + station number

The modified display only applies to incoming/outgoing station numbers. Station number saved for key functions or administered using AMOs are always stored in the format node code + station number.

Prerequisites

The virtual Numbering feature can be activated in HiPath 4000 V 4.0 systems. A regeneration must be performed before activation/deactivation is possible.

Scenarios

- A company with a branch network with a single physical system,
- Shared use of a HiPath 4000 by multiple companies. Firma mit Filialnetz mit einer einzigen physikalischen Anlage

Branch network optimization

Initial scenario:

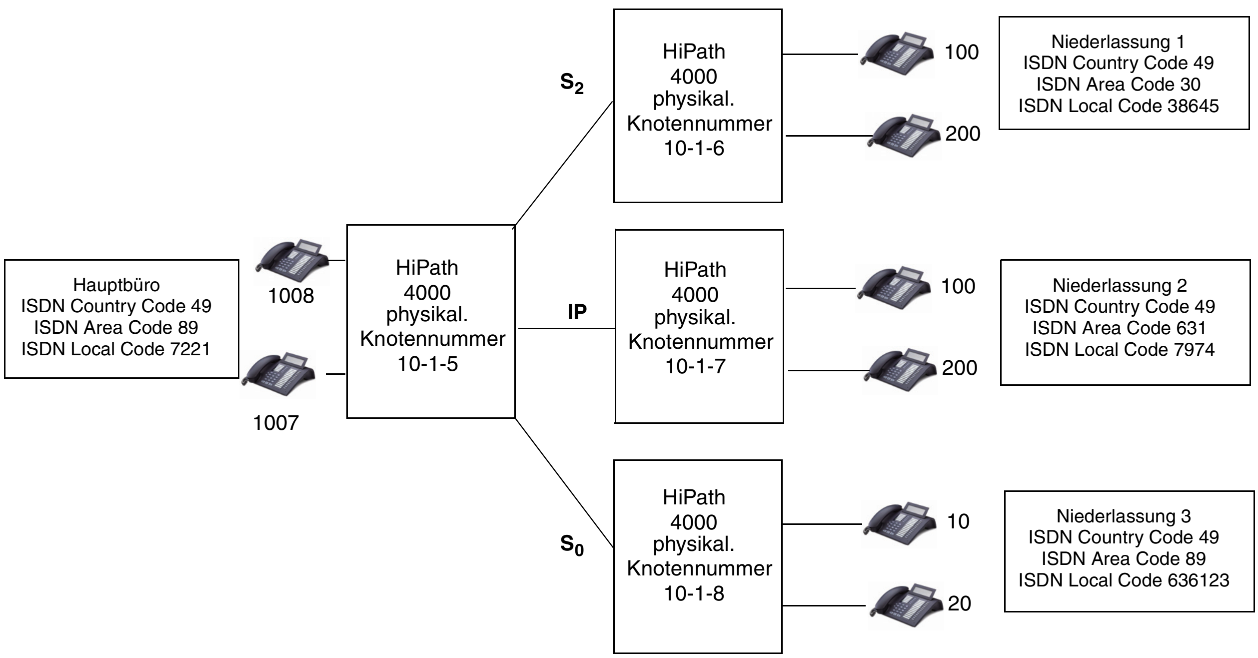


Figure 23 Corporate network with multiple HiPath 4000s and open numbering plan

Target scenario

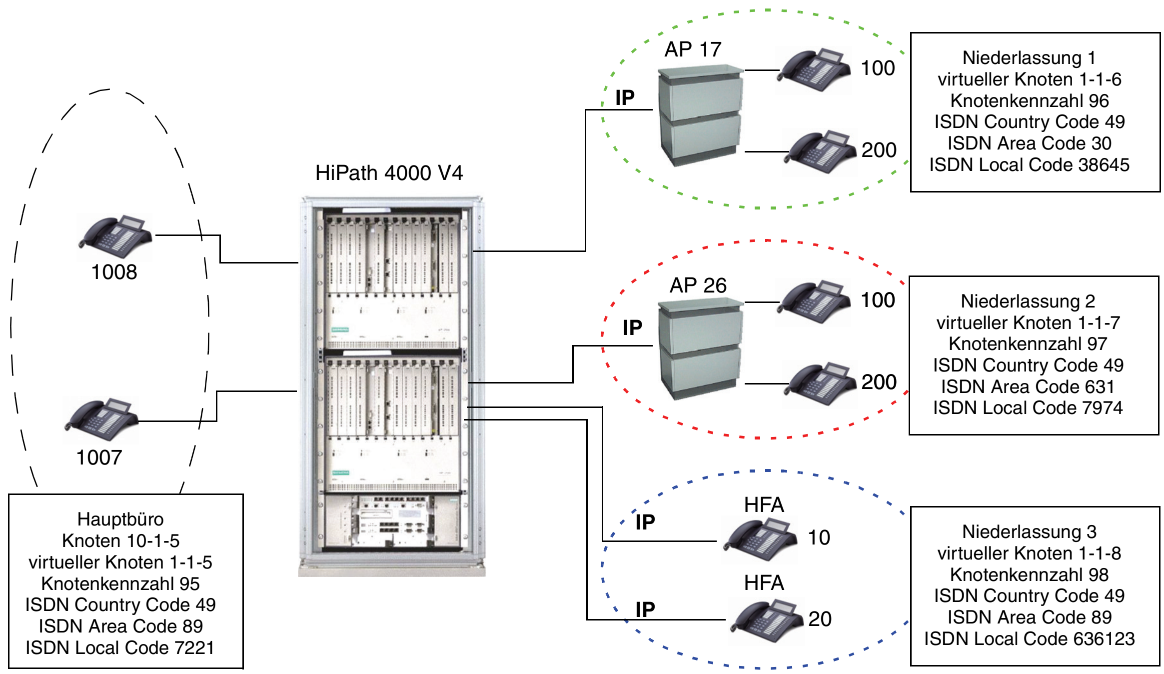


Figure 24 *Identical station structure but with one HiPath 4000, two access points and HFA terminals. With this feature several identical or overlapping numbering plans can be used within a HiPath 4000.*

Shared use of a HiPath 4000 by different companies

Initial scenario

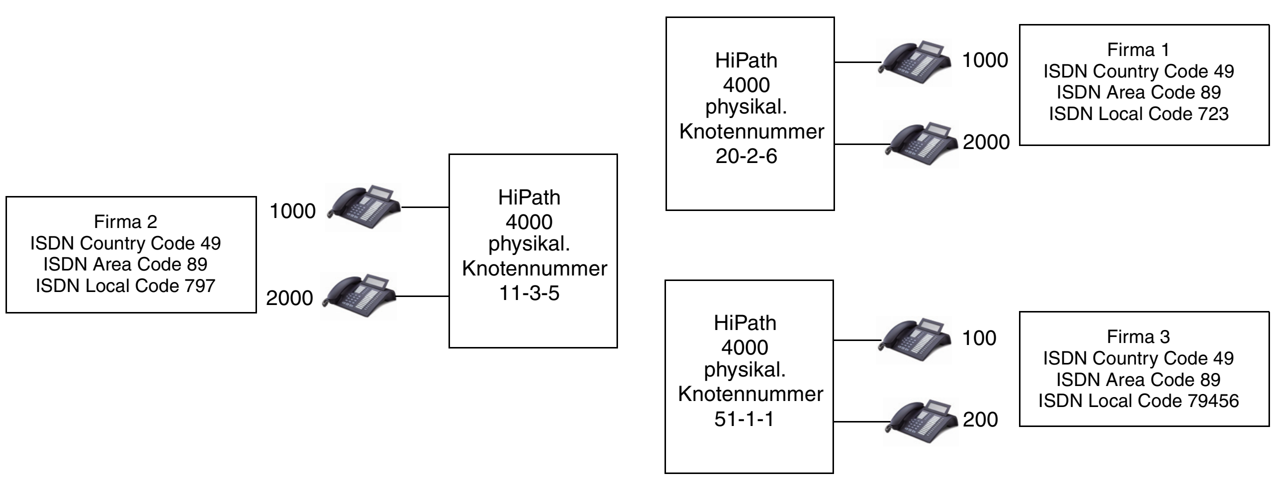


Figure 25 *Three companies with seperat communication systems*

Target scenario

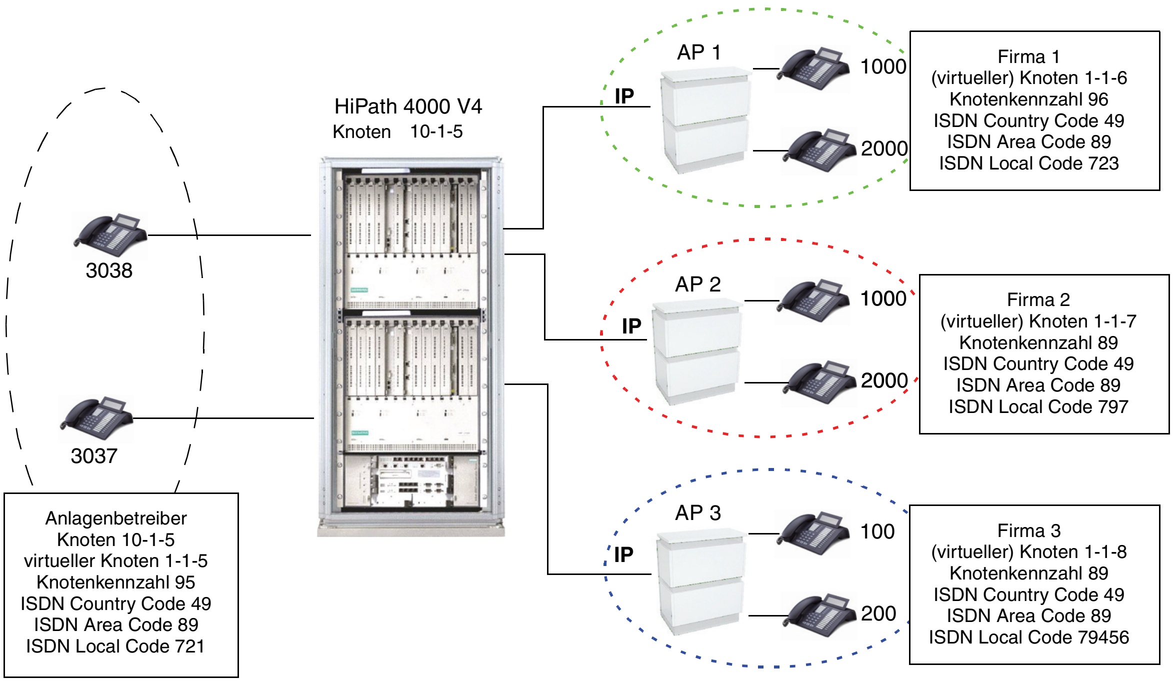


Figure 26 Three different companies in one commen HiPath 4000 system

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IMPORTANT: The virtual nodes are not limitted to access points or host system

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Comments

1. The activation of this feature changes how stations are reached via their station number. This must be taken into consideration when analyzing requirements.
2. When you activate this feature, a station number is only unique if combined with a node code. Station numbers for multiple assignment continue to be configured in the AMO WABE with the digit analysis result STN.
3. Stations can be reached in a virtual node with their station number, even without a node code.
4. Idle terminals display the station number without the node code.
5. The display format for the station number in ringing or call status can be set individually for each customer (international, national, local, unknown). This setting then applies to all stations in the entire physical node.
6. Overlapping station numbers (10, 100, 1000) can be configured on the basis of DPLN groups.
7. similar numbers cannot be in the same WABE group

8. a WABE group may include one or more virtual nodes
9. a virtual node can house subscribers with different WABE groups.
10. Since all subscribers in the branch offices have the same length of digit extension numbers it is possible to use the same WABE group for all virtual nodes.
11. Maximal 16 WABE-Gruppen stehen zur Verfügung
12. A maximum of up to 1000 subscriber groups with their own virtual node are supported.
13. The node code and station number may not exceed 6 characters in length. This yields a maximum phone number length of 12 digits.

Restrictions and Special Features

The following special features arise due to the use of Virtual numbering

- Features, such as, PU group, hunt group, CHESE, direct station selection, and COM group, can be configured for stations in different virtual nodes.
- The node code is needed for unique identification in the Autoset Relocate and Mobile HFA features.
- Autoset Relocate is restricted to subscribers in the same physical node only ACL
- ACL
  External applications using ACL (through CAP/CA4000) always have to identify unique subscribers.
- HiPath Cordless
  If two equal subscriber numbers need to be configured on the same SLC home board the handsets have to register themselves with VNAC+subscriber number. In addition it is necessary to configure extension lines (Verlängerungsverbindungen).

The following restrictions must be observed:

- Keysets with lines over different virtual nodes are not supported. (This restriction also applies to TEAM / CHESE on a network-wide basis.)
- The members of the one number service must be located within a virtual node. Otherwise false information is displayed.
- Closed calling is no longer possible between stations with different E.164 station numbers.
- The virtual node code is a component of the CHRGEE in call data recording.
- Virtual numbering is not supported for feature access codes/function codes and special numbers

- No combination of "7/8-digit numbering for China" and virtual numbering within one node is allowed. The features China 7-digit Numbering Plan and Non unique Numbering Plan exclude each other

4.11 Announcements with Suffix Dialling

In the case of announcements with synchronised text start, the call is held until the text start is signalled to the system by the announcement unit. Only then is the call switched through.

With the aid of the administration and maintenance system, it is now possible to activate announcements for specific destinations. The announcements can be selected and activated according to the following criteria:

- Synchronous or asynchronous announcements
- Announcement reason
- ITR group
- Company name index (if applicable)
- Type of announcement

Synchronous announcements for specific call progress situations in incoming exchange traffic take priority over asynchronous announcements defined for the same destinations. The appropriate lines can be operated with pulse dialling, DTMF or MFC-R2 signalling for initial seizure. However, suffix dialling (signalling down the open line) can only be carried out with DTMF signalling.

- Connecting announcement units via analog subscriber interface: announcements can only be connected in asynchronous mode, i.e. announcements must be kept short, since they are played back in continuous mode.
  - An announcement unit can be connected to an analog subscriber interface, which allows an announcement to be made to each caller before switching through to an alternative station. This can be configured for all user types, but is primarily intended for attendant consoles or hunting groups.
  - The announcement unit is configured as an analog station, and is used for the AVOM or ABA feature (announcement before answering). After a few rings, callers hear a message recorded by the called party. As soon as the message has been played back, the announcement unit sets up a call transfer via flash or ground signal to a user in the same system (home node), or, in the case of networked systems, to a user in a foreign node, provided the nodes are connected via CorNet.NQ or DPNSS1 links. The number of the station to which the call is switched is programmed in the

announcement unit. The announcement unit transfers the call regardless of whether the station is idle or busy. The announcement unit must be set up via AMO so that a re-ring is not possible.

- Connecting announcement devices via TMOM. These announcements are connected in synchronous mode, i.e. they are played back from the beginning in each case. This allows longer announcements to be played back.
  - Announcement devices connected must provide a zero pass signal for the TMOM, so that callers can be switched through to the announcement. If no devices are immediately available, callers are queued momentarily until a zero pass signal from one of the devices indicates that the beginning of the announcement is ready for playback, i.e. the device is available.

  Currently, up to 64 announcements (announcement units) can be configured per HiPath 4000 node, giving a possible total of 64 announcement types. Up to 50 callers can be switched to one announcement device. More than one device can be configured with the same announcement type to accommodate overflow. In extreme cases all 64 announcement devices can be assigned to only one announcement type.
  There are no restrictions with regard to the assignment of announcement devices to announcement types. Each announcement type can be individually assigned to an announcement device, or several announcement types (e.g. sequences) can be assigned to a group or several groups of announcement devices, as required. It is, of course, possible for all announcement devices to be assigned the same announcement type, but not possible for several announcement types to be assigned to one device.

The areas of application for synchronised announcements include:

- Caller prompts for direct inward dialling (DID)
- Caller prompts for dialling selective announcements (DISA)
  This feature is intended for PABXs which operate without DID (e.g. in countries where DID signalling is not possible via the public network).

  Callers receive an announcement, after a defined period. During or after the announcement, callers can enter an internal destination number via DTMF signalling. If a caller does not dial a destination number within a specific time, the call is forwarded to the attendant position (and the caller will hear the appropriate announcement).

  Injection of synchronous announcement is not implemented for DISA applications.
- Caller prompts for suffix-dialling if called line is busy (pseudo-DID)
  Callers will only hear an announcement if the station they have dialled is busy. Any other busy state, e.g. ATB, not authorised, etc. are indicated via the usual tones. The announcement unit is always activated in the PABX with the exchange lines, even if the dialled user is in a different node.

Callers will hear an announcement instead of the busy tone. If the caller dials a different destination via DTMF, the announcement is switched off. If the second destination is also busy, the caller will hear a second announcement. Call charges are, of course, applied as soon as the announcement is played back.

- Caller prompts for suffix-dialling if called user does not answer (pseudo-DID) The announcement unit is always activated in the PABX with the exchange lines, even if the dialled user is in a different node

  Callers receive an announcement, after a defined period. During or after the announcement, callers can enter an internal destination number via DTMF signalling. If a caller does not dial a destination number within a specific time, the call is forwarded to the attendant position (and the caller will hear the appropriate announcement).

- General announcement if called line is busy
- General announcement if called user does not answer
- The Hold-the-line announcement is only injected for synchronous announcements, if the AC does not answer the call. A company greeting is not possible.
- Hold-the-line announcement if attendant does not answer
- Hold-the-line announcement when hunting group number is dialled
- Announcement for automatic camp-on.

4.12 Call Tracing

The tracing feature can be used to record the number of a caller on a printout in case of annoying calls (when calls do not identify themselves).

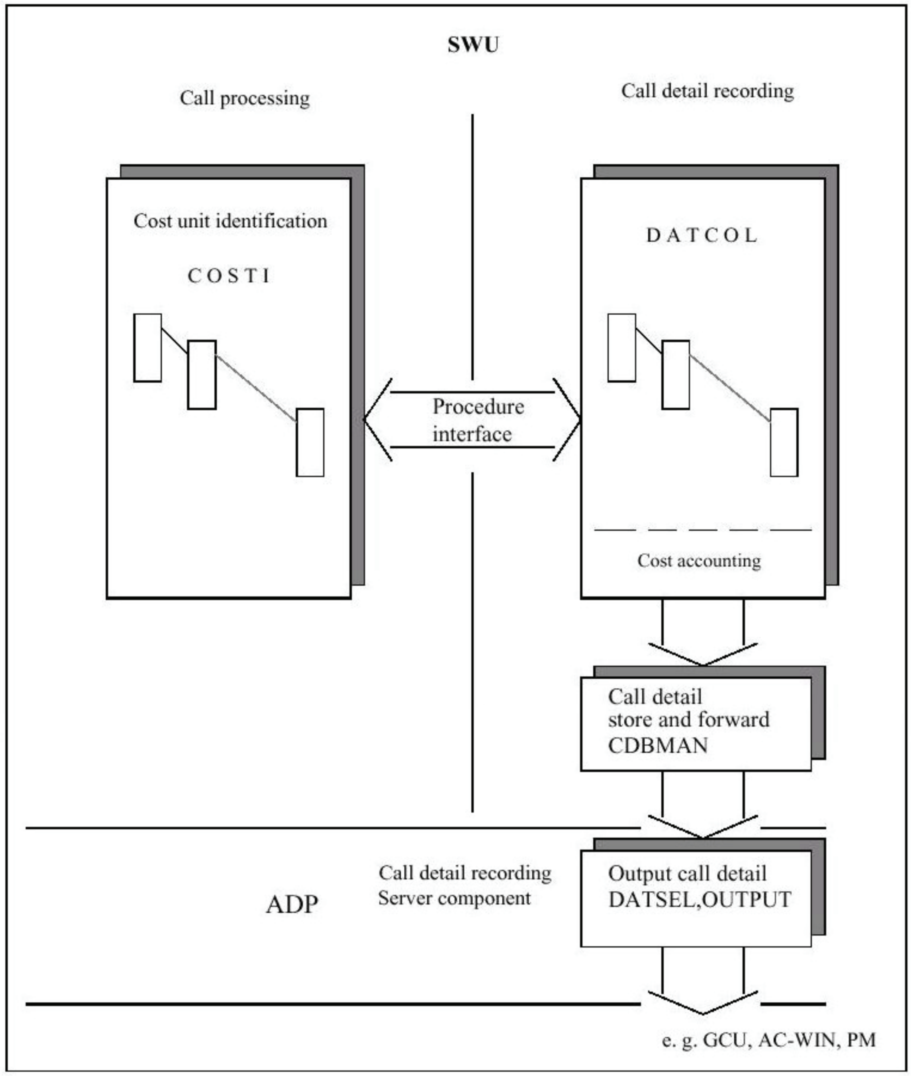
- The system must have the A call number available in order to trace calls (internal call, call from a satellite PABX, private network, ISDN).
- Automatic tracing of all calls (when ringing starts) during a certain period:
  - The administration and maintenance system is used to issue a tracing class-of-service.
  - The numbers of all callers are automatically printed for as long as tracing is activated at a station.
  - The tracing class-of-service can be issued to users with analog-, digital system- or IP telephones, and attendants.
  - The tracing class-of-service can be activated for a maximum of all station users/attendants at the same time.

- Manual call tracing
  - All users can cause the number of the caller to be printed out using a standard procedure.
  - Tracing is performed only for the call for which the procedure was selected.
  - Negative acknowledgment if the feature is not present or the number of station A is not available.
- Printout at a central printer (service terminal)

  For tenant service, printout for all customers at a common printer.

4.13 Call Data Recording

Call data can be recorded and evaluated for every (incoming, outgoing or internal) connection. Call Processing (CP) then transmits the required data of all connections and communication services to the call charge recording.



To implement a uniform modular call charging module, the following steps can be carried out for all communication services:

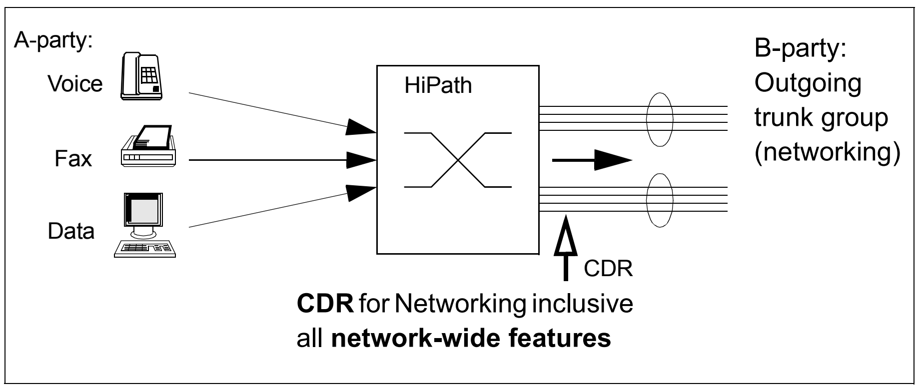
- Collect and edit call charges and call data for each connection (different data, same procedure and format).
- Calculate call charges during the connection for display at the terminal
- Select call charge data: ascertain to which data carrier, in which format and at which tariff output is to be made.
- Record, output, and evaluate call charge data of different types.

4.13.1 Types of Traffic

Call data recording and call charge calculation is realised for different traffic types, and can be enabled or disabled by setting the appropriate feature bits and classes of service (for an entire system, specific lines, or specific users).

Not through-connected connections are also marked in the CDR.

4.13.1.1 outgoing exchange and transit traffic (networking).



Call data recording and call charge calculation can be configured for all types of outgoing exchange and transit (networking) traffic, for the services "voice", "fax" and "data".

Outgoing exchange and network traffic based on direction- and routing dependent cost parameters

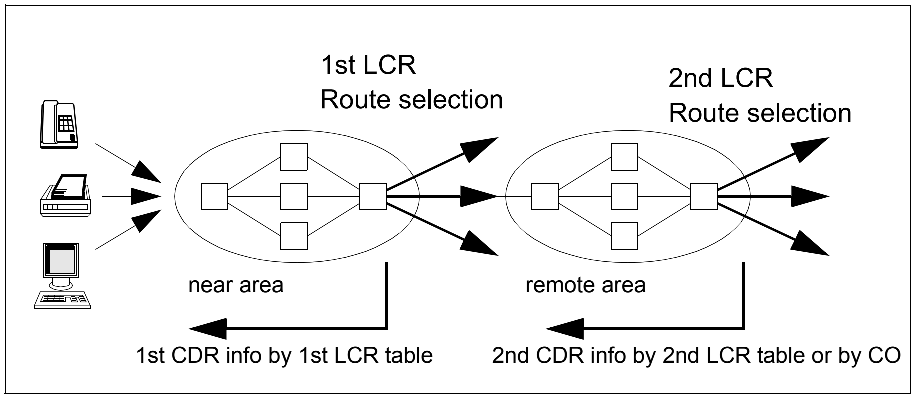
The call charge information for the display is read from the LCR tables, which contain the route-dependent cost parameters (LCR parameters), and is transmitted back through the network to the A-user’s node.

The LCR cost parameter (CARRIER / ZONE) can be assigned to each LCR route element.

This LCR cost parameter is used for the internal call charge calculation of digital system- and IP telephones, as well as for controlling the display output.

Although the call charge information for the display is transmitted back through the network to the A-user’s node, call data recording is always carried out in the node in which the lines/trunk groups of the route element to be recorded are configured.

Call charge information which is transmitted back through the network to the A-user’s node from the LCR node is used for the display. In addition, call charge information of further, subsequent LCR nodes, including call charge information received from a subsequent node where CO breakout takes place, may also be transmitted back to the A-user’s node.



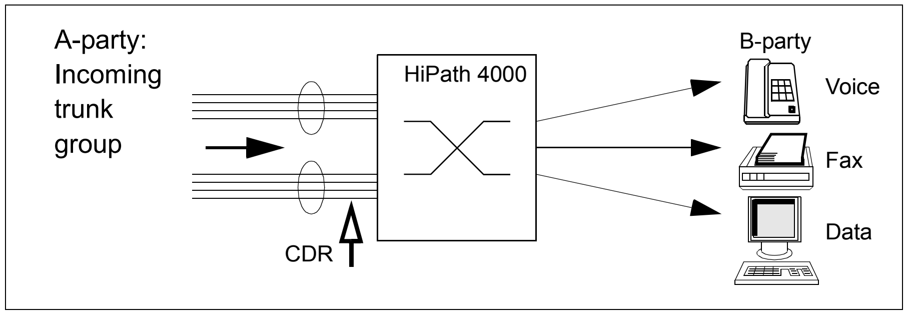
The following variants exist:

Variant 1:
The first CDR information element which reaches the A-user’s node is used for the Digite display. This is the information element received from the first LCR node, in the "near area". Further CDR information elements which reach the A-user’s node originate in the "remote area" following the 1st LCR node, or in the public network, if a subsequent CO breakout takes place. These information elements must be ignored by the A-user’s node for display purposes.

Variant 2:
The first CDR information element which reaches the A-user’s node is received from the first LCR node, in the "near area". The information element read from the LCR tables in the first LCR node informs the A-user’s node that no call charges are to be displayed, since the selected route (or dialled number) is free of charge. This also applies if charges are subsequently contained in the information elements received from the nodes in the "remote area".

Variant 3:
The information element received from the "near area" informs the A-user’s node that no charges are entered in the LCR tables for the route element selected by the LCR system in the 1st LCR node. However, if charges are subsequently contained in the information elements received from the nodes in the "remote area", these must be used for the Digite display.

4.13.1.2 Incoming exchange and transit traffic (networking)



Call data recording and call charge calculation can be configured for all types of incoming exchange and transit (networking) traffic, for the communication services "voice", "fax" and "data".

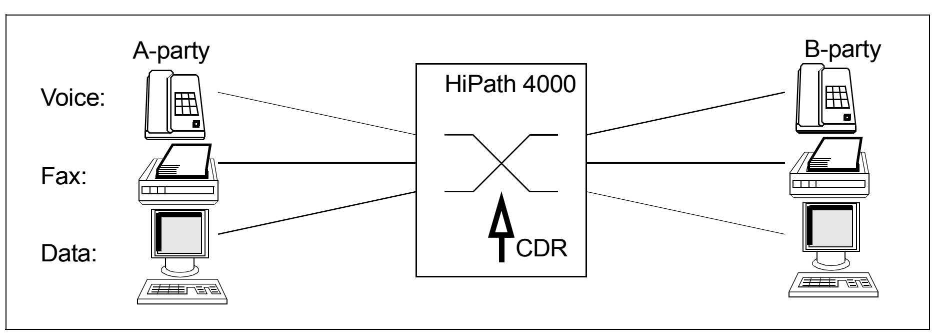
Call charge calculation depends on cost parameters, which can be assigned to each trunk group via AMO.

If AOC (advice-of-charge) information is sent with the connection setup information for incoming Euro-ISDN connections, the information elements are used for call charge calculation.

For incoming analog connections with call charge pulses, these must also be taken into account by the CDR system. However, this does not mean that the CDR feature is released for use with incoming calls from all country-specific analog line variants.

CDR for incoming calls does not mean that the call charges will be accepted ('collect' call feature).

4.13.1.3 Internal traffic



Call data recording and call charge calculation is possible for the features of internal traffic, for the communication services "voice", "fax" and "data" (for the voice service: basic calls, consultation calls, transfer/pickup, three-party conference, toggling, call pickup, call forwarding/forward on no answer, override and callback).

Call data recording of internal traffic is only carried out for successfully established connections and, if ZAND AMO parameter CDRRINGB (call detail recording active with ringback) is set, also for failed connections. Connection setup attempts which fail for any reason (e.g. destination busy, no answer, caller goes on-hook before connection is established) are not recorded.

4.13.1.4 Call Charge Recording Networks

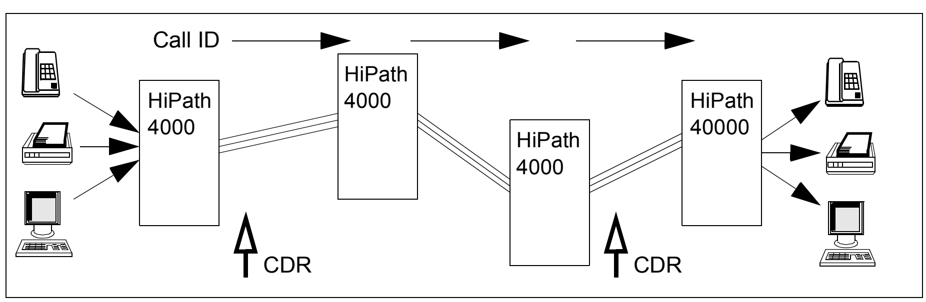
Call charge recording is performed on the basis of all network-wide features. All outgoing
networking connections are recorded, regardless of the feature used to set up this connection.

With this feature, the operation costs incurred by the communication system within the corporate network can be assigned and billed on a station-specific basis.

A inter-network connection is identified with a Call ID and stored with this Call ID in the call charge records.

Network connections can be identified with the aid of a call-ID

This allows call data records for the same connection, which were recorded in the different nodes with the various route elements of the connection, to be identified for later off-line evaluation and call charge calculation.

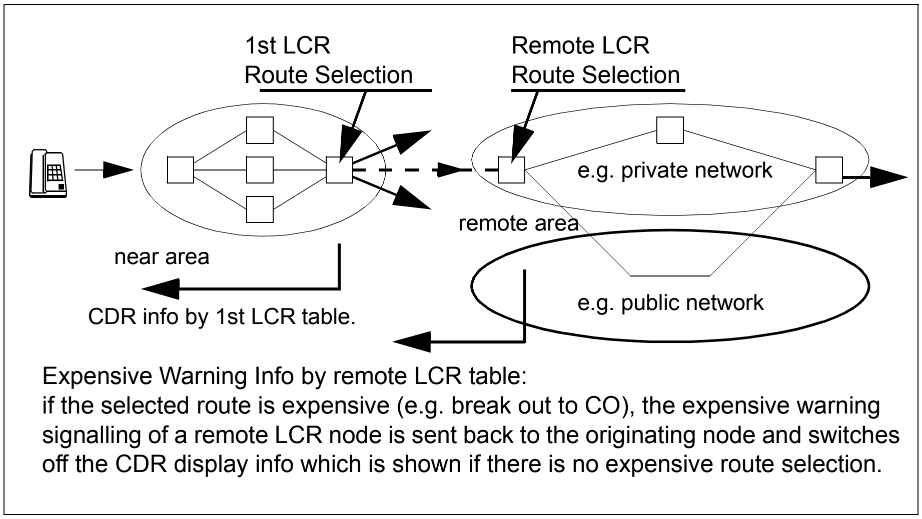


4.13.2 LCR Expensive Route Warning -

LCR expensive route signal if LCR system selects expensive route.

For "normal" routes, the "normal" call data will be displayed. If a route element is selected in a remote node, which requires that additional call data is sent to the originating node (e.g. CO breakout), the LCR expensive route warning function prevents this call data from changing the call data display in the originating node.

In addition, the "expensive warning" signal is registered in the call data record of the connection concerned (field: EXPENS).



4.13.3 Call Charge Calculation

A call data recording and call charge calculation should reconstruct the computations of the postal administrations and network carriers. The methods currently used for computing connection charges differ world-wide, but can be summarised as follows:

1. Clock-based call charge units sent to the PBAX by the public exchange or network carrier during or immediately after the connection:
   - Charge pulses (50Hz, 12kHz, 16kHz), in analog networks
   - Advice-of-charge information elements (AOC-D, AOC-E) as call charge units or currency amounts, in ISDN networks
2. Time-based call charge units sent to the PABX by the public exchange or network carrier at the start of or during the connection:
   - Price per time unit (AOC-S)
3. No charge information elements sent to the PABX by the public exchange or network carrier. Charges are calculated on the basis of internal tariff tables, where the price per time unit depends on:
   - Zone

– Connection time (duration)

– Tariff rate according to time-of-day/holiday.

– Number of used B channels

4. Volume of data and a sliding scale of charges depending on the time of day within a charge period are not supported.

These types of call charge calculation may also be required in combination, (e.g. charge information may be provided for long distance calls, but not for local calls).

The decision, which type of call charge calculation and/or which tariff tables to use for a specific connection type depends on the following factors:

- Carrier parameter entered in LCR tables for route element of connection.
  The LCR tables contain Carrier and Zone parameter values for each possible route element. These values are transmitted to the CDR system each time a connection is set up.
- Tariff group.
  This table contains the numbers of the assigned tariff tables per carrier and communication service.
  6 tariff group tables are defined.

4.13.3.1 Call Charge Calculation with Charge Units or Currency Amount (Charge Pulses, AOC-D, AOC-E)

Call charge calculation is based on the number of call charge units or currency amounts sent by the carrier switch to the system per connection.

The number and time of pulses output depends on the exchange or country and can take place during the setup phase in the exchange, when the destination station answers, during the connection, after release (up to about 2 s) of the connection (advice of charge at end of call).

Pulse transmission is implemented by the exchange via the speech path by injecting 16-kHz pulses, 12-kHz pulses, or 50-Hz pulses. AOC-D and AOC-E information elements are transmitted by the ISDN exchange via the D-channel.

Tariff tables with call charge information (units or currency amount):
The tariffs for connections with call charge information are divided into tariff subunits for calculating the charges on a sliding scale. Each tariff subunit has a call charge limit value, up to which the tariff of that particular subunit applies. All tariff tables are set up with 4 tariff subunits, a minimum charge and a multiplication factor for currency amounts
All "prices" (rates) are specified in one thousandths of the valid currency unit (corresponds to lowest-value currency denomination).
The multiplication factor is only used when carriers supply charge information in

the form of full currency amounts (e.g. US $ 2.75). In this case, the currency amounts must be multiplied with the multiplication factor 100 and the minimum charge does not apply.

Currency specification is part of output formatting. The decimal point position can be defined specifically for each customer.

If a connection is split into different call segments by the call processing system of the system, the associated charges are recorded separately for each segment, and assigned to the appropriate chargees.
In the case of connections with advice-of-charge at the end of the call (AOC-E), the chargee of the first call segment is always charged.

Charges (charge pulses) incurred prior to extension by an attendant are always assigned to the chargee of the call segment immediately following call acceptance.

4.13.3.2 Call Charge Calculation with Time-Based Charge Information (AOC-S)

Call charge calculation is based on the AOC-S information elements sent by the central office/carrier switch to the system per connection.

These are transmitted:

- When the called subscriber/user answers (setup)
- During the connection

AOC-S information elements are sent by the ISDN exchange via the D-channel.

The charges are calculated on the basis of the duration of a connection and the type of charge information received (price per time unit or flat rate). Since AOC-S information elements can be received several times during a connection, the charges are calculated at the moment the new AOC-S information element is received, and stored. This is repeated as often as the AOC-S information is received, or until the connection is ended.
If the AOC-S information contains more than one field value, i.e. more than one service is being used, the charges are calculated separately for each service.

An AOC-S information element can contain the following information types:

- DURATION: this type contains the ’price per time unit’. The connection time is divided into the appropriate time unit value and the result multiplied by the ’price’.
- FLAT_RATE: this type defines a fixed cost (full currency amount) for the connection, independent of time and zone.
- VOLUME: this type is ignored, since volume-dependent tariff calculation is mot implemented.

- SPECIAL_CODE: this type is also ignored.

The currency and multiplier factors are calculated according to the appropriate currency unit and comma position set in the system with the DAVF AMO.

If the call charge calculation is required to result in call charge units, but the AOC-S information is supplied in the form of a full currency amount, these can also be calculated from the given charge amount.

If a connection is split into different call segments by the call processing system of the system, the associated charges are recorded separately for each segment, and assigned to the appropriate chargees.
In the case of connections with advice-of-charge at the end of the call (AOC-E), the chargee of the first call segment is always charged.

Charges (charge pulses) incurred prior to extension by an attendant are always assigned to the chargee of the call segment immediately following call acceptance.

4.13.3.3 Call Charge Calculation without Charge Information from the Exchange or Carrier

In these cases costs are computed on the basis of the duration of a connection, produced exchange-specifically by measuring the time between the following times and the end of the connection (release):

- Seizure of the exchange circuit
- End of dialling in the system
- Answering by the destination station

Only successful internal connections are recorded (answering of the destination number).

There are 9 tariff tables with call charge information.
For connections without call charge information, the tariff rates are entered dependent on time and zone.

Up to 5 tariff periods can be assigned for each tariff or "distance" zone.: "Day", "Night1", "Night2", "Night3", and "Night4" tariff rates per time pulse. The connection cost will be calculated on the basis of the tariff periods.
A basic charge and a minimum charge can be defined.
The basic charge can be entered as a positive value or as a negative value. A negative value is only used if the first n seconds of a connection are free. As long as the calculated connection cost is a negative value, the display output will show COST=0.

The tariff rate periods are assigned in the so-called "Segmentation Tables for Time of Day " (with the TABT AMO). Up to 25 segmentation tables with 24 time-of-day periods each can be defined

Holidays can be equated with any weekday (e.g. Sunday).

The code for tariff rate period changeover specifies whether or not a changeover is to be taken into account when the connection costs are calculated.

The calculated cost of a connection depends on the number of time pulses metered, the tariff rate period and the tariff zone. The number of time pulses metered is multiplied by the tariff found in the appropriate tariff table. The basic connection charge is added to the sum total thus obtained. If the calculated connection cost is greater than 0 (i.e. a positive value), but less than the minimum charge, the minimum charge is applied instead.

If a connection is split into different call segments by the call processing system of the PABX, the associated charges are recorded separately for each segment, and assigned to the appropriate chargees.
In the case of connections with advice-of-charge at the end of the call (AOC-E), the chargee of the first call segment is always charged.

Charges (charge pulses) incurred prior to extension by an attendant are always assigned to the chargee of the call segment immediately following call acceptance.

4.13.3.4 Selection of Call Charge Data

By comparing the call data collected in the standard data record with the customer-specific selection table, decisions are made as to whether the data is to be output at all and if so then to which data media (basic or standby device), with which blocking, in which format and with which tariff.

There is a maximum of 4 types of output device (i.e. a maximum of 4 destinations such as printers, hard disk files, attendant console). An output device represents a selection group consisting of 8 selection tables. Four selection groups are possible, i.e. there are 4x8 selection tables. The 8 selection tables per group are checked serially. If a connection attribute match is found in a selection table (selection criteria are set customer-specifically per table), the remaining tables are not checked. It is possible to set output/storage for each connection attribute to which a match is found. Conversely, it is also possible to suppress output/ storage specifically, as required. If no match is found in any of the 8 tables, then output/storage does not take place.

If the data tallies with the parameters of the selection tables in different selection groups, multiple output to different output devices can take place.

If the basic output device cannot be accessed for output (e.g HD cannot be accessed due to buffer overflow, or printer/data link is defective), output can be switched to the reserve device. If output to the reserve device is not possible either, a high-priority message is triggered which leads to an alarm.

Due to the amount of data to be processed for internal connections, it can be necessary to configure immediate output to an output device. In order to do this, you must set an appropriate bit ("output on fast device") in the call data recording bit string with the FEACG AMO. If this bit is set, call data selection will only be carried out with the first selection group. The remaining selection groups are deactivated. For this type of output, you can also assign a basic device and a standby device.

Possible (feasible) combinations e.g.:

- DPS and FT-file (File transfer file)

The data records cannot be alternatively stored for fast output on demand (CDRC file).

4.13.4 Call Charge Data Collection

The data needed for call data recording occur in different time phases during a connection. It is registered for each trunk circuit/line by the CP by means of the charge reception equipment (CRE) and immediately passed on to the call charging module, which collects it for each call until the end of the call. During a call, the call charges incurred are calculated and passed on to display output.

CP/call charging module interface

- The interface is used by the CP to output the call data information relating to external connections which is needed to implement data editing in the call charging module for all types of call data recording.
- The output is so defined as to time and content that it can be used for all communication services.
- Outputs are mainly needed at times which are important for call data recording in different phases of external connection setup or in internal/ external status changes resulting from call processing procedures. These include, for example, start and end of the connection, change of chargee, call ’segments’.
  Call data recording of internal traffic is only carried out for successfully established connections.

Outputs are made at the following times or initiated by the following call processing events:

1. Seizure
   - Outgoing external connections: after seizure of a free external circuit (exchange trunk) or tie trunk circuit (tie trunk in the network) by system-specific (exchange code number) or communication-specific exchange request by
     – authorised main PABX station,

- authorised satellite PABX station via a main PABX circuit,
- attendant for setup by the station or with assignment,

- Incoming external connections: after incoming seizure of a free external circuit (exchange trunk) or tie trunk circuit (tie trunk in the network).
- Internal connections: no output.

2. End-of-dialling
   - Outgoing external connections: after the complete destination number has been dialled to the exchange, irrespective of the internal signalling method, and as soon as charge information is received, by
     - Timeout
     - Evaluation of number of digits (internal end of dialling)
     - Reaching the dial memory capacity (internal)
     - End-of-dial criterion from the exchange

     The end-of-dial criterion is omitted, if the signalling method includes a through-connect criterion with which charge recording starts.
   - Incoming external connections: no output.
   - Internal connections: no output.
3. Answering
   - Outgoing external connections: after analysis of exchange criterion and communication service indicating that the dialled connection has been set up. This is the case when the following arrive:
     - first call charge pulse,
     - answering criterion,
     - through-connect criterion.
   - Incoming external connections:
     - B-user answers,
     - Answering criterion received after forwarding to external destination
   - Internal connections: B-user answers
4. Call segment
   - Every change of originator/chargee (user or attendant) is registered.
     - The criterion for change of call charge assignment is the moment at which the new chargee accepts the costs of the call

   – With each change of originator/chargee, the new call segment is allocated a new call ID.

- Change of originator/chargee for multi-address messages
- "Virtual" call segment (not initiated by CP)
  When certain limit or "threshold" values are reached, a "virtual" call segment is recorded. This initiates an output of the call segment data even if the originator/chargee remains the same, so that the fact that a limit value has been exceeded is also recorded.
  The following limit values can be defined for up to six specific route codes (and the remaining route codes as a group):
  – Charge units,
  – Connection time,
  – Charge amount (currency amount).

  In addition, "virtual" call segments can be recorded for all ongoing connections at a specific time of day.

  The limit values are defined with the GRZW AMO (see AMO description in Service Manual).

5. End

   Release after outgoing seizure of an external or tie trunk circuit irrespective of whether connection setup was interrupted or the connection was chargeable, taking into account the call charge criteria to be expected in certain communication services, e.g. call charge pulses after the end of the connection.

6. Call charge units (not initiated by CP)

   - The call charge units received during a call are transferred to the call charge calculation module via a software interface. The calculated call charge amount is then transferred back to CP for display purposes.
   - For connections without call charge units, the call charge amount is calculated periodically, according to the connection time, and transferred to CP for display purposes.

4.13.5 Output of CDR Data

4.13.5.1 Individual Call Data Output on Telephone Display (CDRSTN)

Output of current sum of call charges on display of digital system- or IP telephones, provided the user has the CDRSTN classmark (call data recording at station).

Instead of currency amounts, call charge units can be displayed. The charge unit multiplication factor can be defined separately per user.

The display is updated periodically (via a central PABX timer), or continuously with the arrival of each charge pulse from the exchange during the connection, and at the end of the connection, regardless of whether the connection was set up by the user himself or extended by another user or the attendant.

When a connection is set up, or if an extended call is accepted, the counter is reset to zero.
If a connection extended by the attendant is accepted, the charges incurred prior to the user accepting the call are assigned to the user, i.e. the display will already show the appropriate amount.
If a user sets up a chargeable connection while already engaged in a chargeable connection (e.g. consultation call), the charges of both connections are added and the sum is displayed on the user’s telephone. However, "foreground" calls and "background" calls are treated individually, i.e. the call charges are displayed separately in each case, and not added.

The charges are transmitted by the HiPath system to a Euro-ISDN terminal during and at the end of a connection, but not immediately on connection setup.

The dialled number display is cleared as soon as the first charge information is received by the terminal.

When the connection is released, the charge display remains for a further 5 seconds before being reset (timeout).

Call charges are output on the user’s telephone display in the following cases:

- Call charge display for internal connections.

  The display shows the running charges calculated on the basis of the duration of the call and a cost parameter (can be internally configured per ITR group). The display output is updated periodically. The display timer is centrally configured in the communication server via AMO.

- Call charge display for incoming exchange and tie-traffic.

  The display shows the running charges calculated on the basis of a cost parameter assigned per trunk group via AMO.
  If AOC-S, AOC-D, or AOC-E information elements are transmitted for

incoming ISDN connections, these are multiplied with internal cost factors, which can be configured separately in each case.
If call charge pulses are transmitted on analog lines for incoming connections, these are also taken into account for the display output as a kind of interim information. This does not mean that the call data display function is enabled for all incoming analog connection types, especially if these follow country-specific parameters.
The display output is updated periodically. The display timer is centrally configured in the system via AMO (see Call charge display for internal connections.).

- Call charge display (CDRSTN) for network connections, based on route-dependent cost parameters (LCR parameters).

  The display shows the running charges calculated on the basis of the duration of the call and a cost parameter assigned per LCR route element (trunk group) via AMO.
  The LCR Expensive Warning signal is used to suppress the display output of "wrong" call charge data through "expensive" route selection in a remote node. This means that for "normal" routes, the "normal" call charges will be displayed, and that "expensive" routes selected in remote nodes (e.g. CO break-out) are ignored.
  The display output is updated periodically. The display timer is centrally configured in each system via AMO (see Call charge display for internal connections.).

- Call charge display (CDRSTN) for Euro-ISDN exchange line connections.

  The charges are displayed immediately following receipt of the AOC-S charge information (advice of charge - setup) from the ISDN exchange. The charge information elements received are multiplied with internal cost factors, which can be configured for Euro-ISDN exchange lines.
  The display output is updated periodically. The display timer is centrally configured in the system via AMO (see Call charge display for internal connections.).

4.13.5.2 Call Data Display per Connection at the Attendant Console (CDRATND)

With the CDRATND feature, call charge and connection data of chargeable outgoing trunk calls are selectively registered immediately after the end of the connection or call segment and shown on the attendant console display with/without call charge calculation (amount).

Users whose calls are to be displayed at the attendant console are assigned a CDRATND classmark (for voice and fax connections).

Call data display at the attendant console can be enabled for the following situations:

- For all chargeable connections.
- For chargeable connections extended by an attendant. The R or P key on the attendant console is pressed during connection setup and extension; all subsequent call segments are displayed regardless of CDRATND classmark and limit value.
- For chargeable connections set up by users with the appropriate classmark. The customer-specific selection criteria in the appropriate selection group are taken into account.

Call charge display at the attendant console is carried out as follows:

- If calls are set up by the attendant, the charges are displayed at the attendant console at which they were set up.
- If calls set up by a user with the CDRATND classmark, the charges are displayed at any attendant console with the "display at attendant console" operating mode within the attendant group responsible for the user.

Indication of a call with call charge display: LED on GA key flashes rapidly

Interrogation of a call with call charge display: Attendant presses the GA key

Call charge display

- Call number of station (max. 22 digits)
- External call number (max. 20 digits)
- Call charge units (max. 4 digits, max. 4095 call charge units) or
- Currency amount (max. 7 digits)
- Duration of call (max. 6 digits)

Suppression or partial suppression of the destination call number: The last n digits of the destination call number are suppressed in the display and blanks are output instead.

Call charge display for attendant connections
(via R or P key; without assignment/extension)

The charges incurred by the attendant or night station are always charged to the user to whom the connection is assigned or extended.
For local connections with advice of charge at the end of the connection, the user who originally set up the connection (first user) is charged.

CDRATND can be used as an alternative to printer output in the event of printer failure.

4.13.5.3 Call Data Display at the Night Station

- The night station must be assigned the CDRSTN feature for display output of call charges on digital system- and IP telephones for connections set up at the night station (voice and fax only).
- For CDRATND output during night service status, a printer can be configured for the night station.The printer can be operated in parallel with the printer at the attendant console (or alternatively).
- CDRATND output can also be configured for the night station display during night service status.
- If CDRATND output during night service status is not configured, (no call charge output at night station), the data is stored until the first attendant console with the CDRATND feature is in operation again. The data is stored until the limit value of the CDRATND memory is reached.
  The limit value depends on whether further output devices are available.

4.13.5.4 Immediate Call Data Output to External Devices (CDRATND, CDRC)

IMPORTANT: This section also describes the CDRATND feature with output to printer but not CDRATND with attendant console display.

Call charge data with or without computed costs can be stored on one or more external data storage devices for later off-line processing.

For all communication services and calls, output is in chronological order immediately after the end of the call or call segment, i.e. without explicit CDRC file output.

Output device types (up to 4 devices available in the system)

- Service terminal (Service-PC)
- 2 CDRC files on PABX hard disk for later processing
- FT file on hard disk for later file transfer
- On-line data transfer to an external computer

Operating modes for charge and call data output

- Single mode (selection)
- Parallel mode
  Exception: parallel mode with CDRATND for attendant console display is not possible

- Standby operation (emergency operation) if the basic unit fails. The system switches to the standby unit when the memory capacity of the call charge data buffer has been reached, or immediately on failure of the main unit (configured via AMO).

Output data can be defined customer-specifically for each format:

Automatic format changes

Suppression or partial suppression of the destination number
The last "n" digits of the destination call number are suppressed during output and blanks are output instead.

The administration and maintenance system (highest access level, only for vendor) can optionally be used to prevent storage of the last two digits of the destination call number for purposes of call data recording. Blanks are stored instead of the two digits.

The automatic format changes can be applied customer-specifically according to the selection criteria.

Call data editing; handling call segments

Proportional cost distribution for charges, currency units or call duration, resulting from the following call processing procedures:

- Transfer (station → station)
- Transfer (attendant console → attendant console)
- Station transfer security (station → attendant console)
- In assignment/extension (attendant console → station) the costs are always charged to the destination user.

Call segments independent of call processing procedures

On reaching a customer-specific, defined limit or the capacity limit.

- Up to 6 limit values are possible, depending on the route code. Reaching one of these limits counts as a call segment.
- Daily, at a predefined time, a call segment is created for all established calls.
- Limit values for call units, amounts and connection times can also be configured for different carriers (max. 9). The lower limit value (derived from the routing code/carrier entry) is relevant for the evaluation of a connection.

Call segments in calls with PIN entry (voice, facsimile)

- For data recording after a call or call segment, the personal identification number last entered is output.

   – If no PIN is entered for a new call segment, fill characters are still entered in the output format if a PIN was entered in the preceding segment. However, this does not apply to "virtual" segments defined by limit values.

For call segments of external calls with call charge information after release, the first user is always charged (voice, facsimile, videotex).

4.13.5.5 Call Data Recording with Output on Request

Call Data Output without Analysis to External Devices/Computer on Request

Output of edited individual call data processed in accordance with CDRC features and stored as hard disk files in a long-term memory.

Filing method (in long term memory)

- Unsorted and unselected in chronological order,
- Unsorted and selected in chronological order.
  Selection features as for CDRC

Output of

- Complete file,
- Part files (time parameters).

Output device

- As for CDRC, except attendant console
- Call charge computing universal GCU stage 3 for networks

Output format

- As for CDRC

Output data

- As for CDRC

Dialog devices for retrieval

- Printer terminal (service terminal)
- Data transfer from the data processing system

Time of data output

- Immediately after request:
  – Manually
  – Automatically by data processing system

– Automatically with customer-specific timing jobs (Up to 8 timing jobs; one-time or periodic tasks)

If the memory function is faulty the system switches automatically to immediate output on the standby device.

4.13.5.6 Call Data Output in FT File on Request

Output of edited individual call data processed in accordance with CDRC features and stored as FT file.

Filing method (in long term memory)

- Unsorted and selected in chronological order.
- Selection features as for CDRC

Output of complete file through File Transfer

- File transfer to data processing system or HiPath 4000 Manger COL V4 for networks

Output format

- As defined for FT file output

Time of output

The call data records are stored in the FT file in the required customer-specific format or COL format. The data record format can be defined via AMO. The FT file can be closed and immediately copied to a predefined file name manually or automatically at a specified time. This is the file which can then be transferred. Once the file has been transferred, it must be deleted from the source hard disk (this is not automatically done by the CDR system).

If the file still exists when the next file is to be renamed, the new file contents are then simply appended to the old file. If this file, in turn, is also not deleted after transfer, then the new call data records will continue to be appended to the old file until the storage space on the HD is exhausted, after which data will be lost (overwritten).

The FT file is stored either during immediate output by the selection or during downloading the call data from the GEZ file into the FT file.

If the memory function is faulty the system switches automatically to immediate output on the standby device.

4.13.6 CDR Account Code

The Account Code (ACC) is used to bill call cost and time to individual customers or projects. The user can add an ACC to an incoming, internal and outgoing call. The activation of ACC is possible before the call or during the call with feature code or ACC-key or via the menu. The ACC is valid only for the current call. But, when that call is transferred to another station, the ACC is still valid.
During the call, it is possible to use more than one ACC. In this case, each ACC is transferred to the charge data recording (CDR). But, in the case of call transfer the latest ACC is used.
The extensions can be configured also for automatic transmission of an ACC on every originating call. The automatic ACC can be overwritten by a manually ACC.

4.13.6.1 Feature

The following is the implementation principle for activation before and during the call:

- FC-bd = Feature Code for ACC-feature-activation before or during the call
- ACC = ACC entered after FC
- ext# = the subscriber number of the party which uses the ACC feature (enteres the ACC)
- AD's: = Account Data. This term is used here for the summary of data which
  are covered in the existing CDR's and which will not be changed
  for the ACC-feature.
- Call-ID = is the number of the call to which the data above are assoziated.

4.13.6.2 Restrictions

- **Networking:**
  The Account Code is not transferred over a network. That means, the charge data recording has to be in the node of the extension, which uses the ACC.
- **Attendant:**
  CDR Account Code for the attendant is not implemented.
- **Several ACCs:**
  Serveral ACCs in one input procedure is not available.
- **Automatic ACC:**
  The automatic ACC is not available for terminating calls.

- **Devices:**
  Functional devices and Non Voice devices cannot use an ACC. Further for analog terminal adapter ACC is not possible.
- **Dongle:**
  CDR Account Code can be switched on or off with a system parameter.
- **Callback**
  When a user calls a party with ACC and sets a callback at the partner, the ACC does not exist for the callback call from the partner to the user.
- **Last number redial and saved number redial**
  When the user dials an ACC followed by a destination number, the ACC is not stored in the last number redial or saved number

4.13.7 Create Standard Call Charge Data Record

The data collected and selected for each call or call segment are put into standard form in a standard call charge data record containing the basic data necessary for call charge computation and output for all communication services.

The running costs of a call are calculated and output on the telephone display of the digital system- or IP telephone concerned. These calculated call charges are also entered in the CDR data record (DISPCHRG element).

The call data is buffered in a call data pool of the SWU, in order to prevent loss in the case of an ISp failure or the failure of an output device.

In order to minimise a possible loss of data for external (=expensive) calls, the recording of internal calls is stopped when the number of buffer entries reaches 75% of the total capacity of the call data pool. Only call charge records of the (chargeable) external connections are stored. The call charge records for internal connections are lost.

Before this limit is reached, the appropriate error messages/alarm messages are output (at 60%, 70% and 75%).

4.13.8 CDR enhancements (CDRe)

The enhanced CDR can be used for allocation of call costs to the appropiate paying party and also for measurement purposes of the call handling performance of the system.

In addition to the call charges, the CDR output contains information about call duration and intervals, source and destination numbers of the calls, call routings (LCR) etc., which can be transferred to an external billing system .

This ensures that each record contains enough information about every step of a connection (call situation) e.g. the connection is traced precisely in the system, the affected subscriber numbers, affected active features and the correct ringing time for each call processing step are recorded.

All the information for a specific call situation is available in a set of CDR s corresponding to the specific call situation.

For example: Party A has activated CF to Party B and Party B has activated CF to Party C.

With an incoming call to Party A, that is accepted by Party C, one group of call detail records associated with the call is identifiable with an ID and a CDR is produced for each of the separate connection legs.

This feature is inactive as default.

Available Enhancements

The following CDR-enhancements to the existing functionality for incoming / internal calls are realized for all extensions and the attendant console (if not otherwise stated):

1. **General basic call :** A CDR record is output for every step in call progression .

   A CDR record output for busy and unanswered calls with the dialed number and the ringing time is also available.

2. **Abandoned calls to the attendant**:

   - For calls already assigned to an attendant console: If a subscriber calls the attendant and terminates the call before the attendant answers (abandoned call) a CDR record is output. This CDR record contains the dialed number, the ringing time and a sign "call abandoned before answer" or "not_connected " (max 10 characters can be input) and the number of the first reached attendant group number.
   - For queuing calls: This CDR record is also output for abandoned attendent calls when the call was queued in the waiting queue of the attendant group. The queue duration can be displayed in a separate field (steps of 1/10 sec) .

3. **Call forwarding:** CDR records which include the single steps of call forwarding before reaching the final destination are available for all subscriber types and alle types of "call forwarding" (CFNA, CFU, CFB). The ringing time is output per CDR record and not as a total ringing time.

4. **Call Pick-up:** In case of call pick up, at least one CDR record is output, which contains the original dialed number and the actually reached number.

5. **Hunting groups:** The actually reached subscriber number in the hunting group and the pilot number are output in at least one CDR record.

6. **On hold:** This case is marked in the CDR record.
7. **Conference:** In the case of a conference call there is more information output in the CDR record. All connections of this conference are marked in the CDR records.
8. **Ringing and answer time:** It is possible that the ringing time (ringing time is the time until a called party answers) and the connection time are included for the calls' record. The ringing and the connecting time for external (trunk or network calls) connection is always available.
9. **Calls to the attendant:** A CDR record is available for regular incoming calls, for recalls and for intercept to the attendant.
10. **The attendant and the (outgoing / incoming) call assistances:** For the attendant, CDR records are output for incoming, outgoing and internal calls which contain also the ringing time for the transferred calls . Costs for outgoing and incoming / transferred calls from the attendant is optional.

4.14 MLPP (Multi-Level Precendence and Preemption)

The Multi-Level Precedence and Preemption (MLPP) service provides prioritized call handling service. This service has two parts - Precedence and Preemption.

Precedence involves assigning a priority level to a call.

Preemption involves a higher precedence call seizing resources that are in use by a lower precedence call, in the absence of idle resources.

The specific requirements for MLPP are described in the Generic Switching Center Requirements (GSCR) document (September 8, 2003) (please refer to the actual service manual). The GSCR specifies support for different types of switches.

4.14.1 Specific terminology

MLPP Service Domain:

An MLPP service domain consists of a set of MLPP subscribers (MLPP users) and the network and access resources that are in use by that set of MLPP subscribers at any given time. Connections and resources that are in use by MLPP subscribers may only be preempted by higher precedence calls from MLPP subscribers within the same domain. The Defense Switch Network (DSN) service domain is zero (0). Stations and trunks may have their own service domain and preemption can only occur in the same service domain.

Network:

In this standard, "network" refers to all telecommunications equipment that has any part in processing a call or a supplementary service for the user referred to. It may include local exchanges, transit exchanges, and NT2s but does not include the ISDN terminal and is not limited to the "public" network or any other particular set of equipment.

MLPP Call:

An MLPP call is a call that has a precedence level established and is ether being setup or is setup. In Digital Subscriber Signaling System number 1 (DSS1: ISDN Q.931 signaling), an MLPP call is a call from an MLPP subscriber for which a setup has been sent but no DISCONNECT has been sent or received.

Preemptable Circuit:

A preemptable circuit is a circuit that is active with or reserved for an MLPP call:

– (a) within the same domain as the preempting call and
– (b) with a lower precedence than the preempting call.

A busy or reserved circuit for which a precedence level has not been specified is not a preemptable circuit.

Preemption initiating exchange:

A preemption initiating exchange is the exchange that is congested (i.e. no idle circuits) and has received a preempting call setup.

4.14.2 Functionality

4.14.2.1 Originating User

The user activates precedence calls by dialing a code on a per call basis.

1. If the originating user is not authorized for the dialed precedence level or for protected calls, the call attempt will be rejected, the user will be connected to Unauthorized Precedence Announcement and the message "nicht berechtigt"/"Not authorized" will be displayed on the digital system or IP-telephone.
2. If the called MLPP subscriber is busy with a call of the same or higher precedence level or with a call from a different service domain, the originating user will be connected to Blocked Precedence Announcement and the message "bitte wiederholen"/"Please try later" will be displayed on the digital system or IP- telephone.

3. If the called MLPP subscriber is idle, the calling station will receive precedence ringback tone and message "PRECEDENCE CALL" on the display while the called subscriber is ringing. After 4 seconds or when the called subscriber answers the call, the calling station will show the MLPP level in addition to the existing informations (called number, name), if administered.
4. If the called MLPP subscriber is busy with a call of the same or higher precedence level and call waiting for the called party is activated, the originating user will receive precedence ringback tone.
5. If the called MLPP subscriber is assigned as non-preemptible, the calling party will receive "geschuetzt/Protected" and Busy Not Equipped Announcement.
6. If the precedence call reaches an attendant queue, the user will be connected to Attendant Queue Announcement.
7. If the originating user originates a precedence call to a nonexistent number, the user will be connected to Vacant Code Announcement.
8. If technical problems make origination of a precedence call impossible, the user will be connected to Isolated Code Announcement.
9. In case of the DSN Hotline, the user activates the hotline calls by initiating a service request (usually hook-off). The call receives a precedence level automatically, administered earlier. If the called subscriber has a configured Black/White list, it will be checked to see whether the present call is allowed. If not, the originating user will not be connected to the Announcement and the message "bitte wiederholen"/"Please try later" will be displayed on the optiPoint device. In other cases, the procedure is as for simple MLPP calls.

4.14.2.2 Destination User

1. Called MLPP subscriber is idle
   If an MLPP subscriber is idle and is seized with a MLPP call, the display message at the called station will show the MLPP level in addition to the existing informations (calling number, name), if administrated. A special ringing rhythm for precedence calls (one for all levels) may be administered.
2. Called MLPP subscriber is busy (with camp on)
   If an MLPP subscriber is busy with a call of the same or higher level or with a call from a different service domain, during the camp-on phase the display will be the same as in ringing state and the user will hear precedence call waiting tone.
3. Called MLPP subscriber busy (without camp on)
   At the moment of preemption the display will change from the information about "old" connection to an information, that a precedence call is waiting:

"Prioritaetsanruf wartet"/ "Precedence call waiting", and the user will be connected to preemption notification tone. After the called party has gone on-hook and rering starts, the display will be the same as in ringing state.

4. The attendant console is implicitly protected against preemption by precedence calls.

4.14.3 Feature dependencies

4.14.3.1 Precedence Call Diversion

Diversion of unanswered precedence call

1. Originating user calls destination user with precedence call. Destination user is alerting with precedence ringing.
2. Destination user does not answer within a period of time.
3. Call is diverted to the diversion destination.

4.14.3.2 Precedence Call Waiting

Camping on by a lower or equal precedence call or by a call from a different service domain

1. After dialing the access code and the extension number the text "PRECEDENCE CALL" appears on the display of the originating user.
2. The destination user hears precedence call waiting tone.
3. When the destination user goes onhook, it will hear precedence call ringing immediately.

4.14.3.3 Call Forwarding

Unconditional call forwarding at an idle station

1. Originating user dials the precedence level access code followed by the extension number of the destination party
2. The call is forwarded immediately to the call forwarding destination of the destination user. The rules of MLPP will apply to the call forwarding destination.

Preemption at a busy station that has call forwarding activated

1. 1. Originating user dials the precedence level access code followed by the extension number of the destination user. The destination user is busy. It also has unconditional call forwarding activated.
2. The call is not forwarded. The original call of the destination user is preempted.

Call forward on no reply

1. Originating user dials the precedence level access code followed by the extension number of the destination user. The destination user starts hearing precedence ringing.
2. The destination user does not answer. The call is forwarded to a new party. The destination user stops ringing, the new party starts precedence ringing.

4.14.3.4 Conferencing

Preemption of whole conference

When a conference is preempted because of the lack of conference resources, all of its members will receive Preemption Notification Tone. All of the members must go onhook as an acknowledgement.

4.14.3.5 Calls in Queue to the Attendant

Realization of the calls in queue will be done by mapping each precedence level to a specific attendant queue. Each queue has its own key on the attendant console. The administrator has to assign the description text for the different precedence levels to the corresponding call queue keys on the attendant console. The attendant has the ability to get the calls with the higher precedence first by hitting the key for the queue of the higher precedence first. This function is available on AC-Win-MQ only.

4.14.4 Defense Switched Network World Wide Numbering and Dialing Plan

The administrator may configure Defense Switched Network World Wide Numbering and Dialing Plan for

Access Digit, Precedence or Service Digit, Route Code, Area Code, Switch Code and Line Number

Seven-Digit and Ten-Digit Dialing.

4.15 CLI- modification

CLI- modification offers the following feature set:

- incoming external calls can displayed with an internal telephone number. If the incoming telephone number matches a predefined number/pattern, it is changed to an internal telephone number.
- If the incoming telephone number matches a predefined number/pattern, the incoming party name is changed (Alpha tagging)
- Call barring: If the incoming telephone number matches a predefined number/pattern, one of the following actions is taken:
  - the call is rejected
  - the call is routed to a predefined number
  - the call is accepted and routed to the original called party

Restrictions:

- This feature is available on digital central office connections and networking trunks, but should only be used in conjunction with Central Office trunks.
- If an incoming call is rejected by this feature the call is cleared with cause "incompatible destination".
- For an incoming call to a station, who is restricted the hotline destination is called. If this hotline destination is busy the call is cleared with cause "incompatible destination".

5 Features for Subscriber

5.1 Number Redial

Any external or internal call number dialled with the pushbutton set, by speed dialling or with repertory or DSS keys can be stored for subsequent redialling as long as the connection is still alive.

1. For telefax devices, only the call number is stored and repeated, not the telefax code.
2. Maximum number of digits per number redial destination:
   - external destinations: 22 digits including trunk group access code
   - internal destinations: 6 digits
3. All dialling information entered on the pushbutton set, including * and #, can be stored.
4. Every station can store the dialling information for one connection only.
5. Storing the dialled number during an outgoing call (analog telephone)
   - during a first call
   - during a consultation call
     The number redial code must be dialled within 2 seconds after the signal key is pressed or else the consultation call is withdrawn.
   - After the information has been stored (even in call state) the last dialled connection is released.
   - The memory contents are overwritten by the next call number stored by the station user.
   - External call numbers are stored with their trunk group access code.
   - A negative acknowledgment is given if the feature is not present.
6. Storing the dial information during an outgoing call (Digital system-/IP-telephone)
   - During a first call.
   - During a consultation call.
   - Storage of the dial information is carried out while the conversation takes place.
   - The memory contents are overwritten by the next call number stored by the station user.

- External call numbers are stored with their trunk group access code.
- A negative acknowledgment is given if the feature is not present.

7. Storing a call number dialled during the call state (saved number) (Digital system- or IP-telephone)
   - In the call state during incoming and outgoing calls, the station user can enter a call number and store it for later number redial.
   - It is also possible for the station user in idle state to enter a call number and store it for later number redial.
   - Storing of the dial information is carried out while the conversation takes place.
   - Deletion of the dial information is possible by pressing the CL key.
   - The memory contents are overwritten by the next call number stored by the station user.
   - A negative acknowledgment is given if the feature is not present.
   - The dial information is not checked; * and # can also be stored.
8. Storing the call number of a caller (Digital system- or IP-telephone)
   - After the call has been answered, the call number of the caller can be stored for later number redial using the redial key.
   - Store the call number of a first call.
   - Store the call number of a second call (visual call waiting) (optiset E voice terminal).
   - Storing of the call number of the caller is carried out while the conversation is taking place.
   - The memory contents are overwritten by the next call number stored by the station user.
   - Negative acknowledgment
     – If the call number of the caller is not known to the system (outside call, tie-line call).
     – If the feature is not present.
9. The stored number is always the call number of the station user with which the station user storing the number has a connection at that moment. (Analog telephones: outgoing call, Digital system- or IP-telephone: incoming or outgoing call). This is also true if there is a change of call partner during consultation calling or alternating between calls. For outgoing connections the dialled call number is stored. If an outgoing called is dialled using a speed

dialling number as a prefix to the specific user station number, then the speed dialing number is stored and not the converted long number which is used for dial transmission.

10. Storing the dial information in conference state is not possible.
11. Interrogation of the memory contents (Digital system- or IP-telephone)

    The stored internal call numbers (with names, if available) or external call numbers are displayed (external call numbers include the trunk group access code).
12. Deleting the memory contents
    - manual deletion (Digital system- or IP-telephone) using the standard procedure
    - automatic deletion by overwriting the stored dial information
13. Dialling can be repeated as often as desired without renewed manual storing, provided the station user has not stored a new call number.
14. The digits transmitted for number redial of an external call number are subject to toll/code restrictions.
15. When an internal call number is dialled with number redial, the system checks whether the connection is allowed (station–to–station restriction).
16. All station users can use the number redial feature.
17. If a user with a digital system- or IP-telephone in idle state presses the redial key without stored dialling information, a seizure without dialling is carried out. The station user receives internal dial tone.
18. A patch allows the redial function for each individual digital system- or IP-telephone to be set for "last number redial" or "saved number redial". The desired setting can be selected using a specified key configured on the terminal itself.

5.2 Call Forwarding Improvements

5.2.1 Fixed Call Forwarding

User can program, activate, deactivate or interrogate fixed forwarding. Keysets can program forwarding for any line on their phone.

- Programming a destination also activates the forwarding feature and an activated station call forwarding is automatically deleted.
- When the fixed call forwarding is deactivated the destination is not deleted.

- Fixed call forwarding can be activated again by key, access code or menu
- Fixed forwarding is deactivated when station call forwarding is activated
- When station forwarding is deactivated then a fixed call forwarding is not automatically activated again

5.2.2 Station Call Forwarding

Station Call Forward can be programmed and activated by user or by AMO. When deactivated the destination is deleted. Keysets can program station forwarding for any line on their phone.

The forwarding conditions are:

- Station Call Forwarding; unconditional, all calls
- Station Call Forwarding; unconditional, internal call only AND/OR external calls only
- Station Call Forwarding; on busy or no answer, all calls
- Station Call Forwarding; on busy only, all calls
- Station Call Forwarding; on no answer only, all calls

Only one of the listed station forwarding can be activated at a time. Exception: unconditional forwarding for internal and external calls to different destinations simultaneously and

When a different kind of station forwarding is activated the actual forwarding is deactivated and deleted and the new forwarding is activated.

5.2.3 System Call Forwarding

System Call Forwarding can be programmed only by AMO. System forwarding is not deactivated when station forwarding is activated. Station forwarding takes precedence over system forwarding. When station forwarding is deactivated system forwarding is still active.

There exists an access code to deactivate and an access code to activate system call forwarding. These access codes are mainly for service purposes and are usually not told to the user. Use of the access code for deactivation deactivates all system forwardings and code for activation activates all system forwardings again.

- system call forwarding; unconditional, internal AND/OR external calls. This kind of forwarding makes sense only for fictitious devices.
- system call forwarding; on busy, internal calls only.

- system call forwarding; on busy, external calls only.
- system call forwarding; on no answer, internal calls only.
- system call forwarding; on no answer, external calls only.
- system call forwarding; on do not disturb, internal calls only.
- system call forwarding; on do not disturb, external calls only

5.2.4 Forced Call Forwarding (Call Deflection)

When the user hears ringing this call can be forwarded by pressing the CALL FWD or CANCEL key. The call is then forwarded to the ring_no_answer_forwarding destination.

If call forwarding no answer is not possible the key depression is ignored.

Forced call forwarding of a camp-on call by pressing the CALL FWD key is possible.

Note: The CALL FWD key means only forced call forwarding if the device is ringing. If service mode is active it means scrolling and if not it means activation or deactivation of forwarding.

5.2.5 Subscriber control of diversion

This feature allows a user to override a forwarding at the called destination.

5.2.6 General Rules

- Forwarding internal calls include those within the local private network. Forwarding external call include all calls from the public network.
- While call forwarding is activated, the forwarding station remains operational for outgoing calls.
- At any given time a station can have ONE station call forwarding active, and ANY combination of system call forwarding. Station call forwarding will take precedence over system call forwarding, for like forwarding types. Fixed call forwarding takes precedence over other forwarding types.
- Call Forwarding on do not disturb has precedence over call forwarding on busy.

- Forwarding destination number must be terminated with "#" or by selecting the menu option "save?". If "#" is not dialed then the validation of the number starts only after time-out when dialling in the voice function. In the service function there is no time-out.
- Digital system- or IP-telephone without display cannot use the service function to program, activate or deactivate forwarding. They must use access codes.
- Stations with fixed or unconditional station call forwarding activated will receive special dial tone when originating a call, indicating that the "station user" has activated a feature that effects calls terminating to that device. Special dial tone will be synchronous to what the status of the call forwarding LED indicates.
- No special dial tone for system call forwarding!
- No alert tone is heard at the station which has forwarding active when a call is forwarded.
- The forwarded-to station may call the forwarding station and forwarding is ignored.
- Only station call forwarding; unconditional, all calls can be activated or deactivated from a foreign station.
- If station call forwarding; unconditional, all calls is deactivated for fictitious or specially marked stations the fix call forwarding must be activated again.
- When activating, deactivating, deleting or programming forwarding with access codes in the voice function an acknowledgment on the display is given and no confirmation (acknowledgement) tone is connected for digital system- or IP-telephone with display. Phone turns into idle state without user action. Confirmation tone only for digital phones without display.
- US market
  - When activating, deactivating, deleting or programming forwarding with access codes an acknowledgment on the display is given and a confirmation (acknowledgement) tone is connected.
  - Special dial tone is applied if any kind of station forwarding is active.
- Forwarding can be programmed, activated and deactivated from
  - Analog telephone by using access code
  - digital system- or IP-telephone with display by using access code, key or menu
  - digital system- or IP-telephone without display by using access code or key
  - Keysets with display by using access code, key or menu

- Keysets without display by using access code or key
- Functional devices (stimulus) by using access code
- Functional devices (standardized DSS1 diversion) no changes

- Fixed/variable call forwarding - all calls/no answer -
  - To users, servers and paging systems with multiple paging within the same node
  - To users, servers and paging systems with multiple paging at foreign nodes in the HiPath 4000 network, of the public exchange and to users at PABXs behind the public network, subject to the following restrictions:
    - Voice connections only,
    - Only with special classmark,
    - Connection from ISDN exchange to ISDN exchange is possible,
    - Charges for connections from the PABX to the public exchange borne by the forwarder of the call,
- Via administration and maintenance can be set to carry out an availability check on call forwarding destinations if the availability of the call forwarding destination is not checked until call forwarding is set up or that the availability of the destination is checked during programming.
- If variable call forwarding is deactivated and fixed call forwarding simultaneously activated by a "mobile user" via card and PIN at a "foreign" station, no checking is carried out. This is because the PIN card user would otherwise not be reachable if the fixed call forwarding destination were to go out of service for even a short period.
- Calls to stations with call forwarding, whether in the free or busy state, are immediately forwarded to the destination station.
- With a digital system- or IP-telephone, the number of a call forwarding destination (preset or variable) can be entered by pressing a repertory or DSS key instead of using the pushbutton set.
- Call forwarding to the attendant console
  - Call forwarding to any AC. The station user enters the attendant code as the destination data.
  - Call forwarding to a specific AC. The station user enters the AC call number.
- Prevention of call forwarding,
  - when the call station (dialled destination number) does not have the class-of-service necessary for the call (trunk, tie-line or DID access),

    - when the connection between the caller and the called station is not permitted (station-to-station restriction),
    - when the connection between the caller and the destination station reached by means of call forwarding is not permitted (station-to-station restriction). This restriction is optional and can be activated for each system with the aid of the administration and maintenance system.
    - In such cases the caller receives an appropriate notification.
- Call forwarding is ignored (call is signalled at the called station)
    - if the call is from the destination station,
    - if it is a callback (always signalled at the initiating station),
    - if it is a recall because the called station went on-hook during consultation,
    - if it is a DSS call,
    - for calls switched to night service (night service with activated call forwarding),
- If call forwarding is activated for a station (with the exception of the pilot station) in a hunting group, it applies to personal calls only. Hunting group calls will continue to be signalled at the station.
- Irrespective of the type of call forwarding initiated (preset, variable, follow-me), by means of the administration and maintenance facilities it is possible to specify whether calls are forwarded once only (single-stage) or "chained" (two-stage).
    - With single-stage call forwarding, the forwarding destination is called even if call forwarding - all calls or call forwarding - no answer is activated there.
    - With two-stage call forwarding, "chaining" is possible for voice connections only, if the first call forwarding destination is a station line, irrespective of whether it is accessed in the same system as the station originally called or by way of networking
- In addition to the standard variable FWD functions , multiple or linked call forwarding is possible ("Follow-Me" feature). The number of Follow-Me steps can be configured from 1 to 10 destinations on a network-wide basis, and again on a system-specific basis.
  Until the final version of the feature is released, the following restrictions apply: the feature is only implemented for the VOICE (VCE) communication service. The DATA (DTE) and FAX services are not supported.
- It is possible to prohibit FWD via the public network when multiple FWD has been set, unless the user forwarding calls via the public network is called directly. This functions with the following restriction: The user forwarding calls

via the public network has to be configured in a node with direct access to the public network, i.e. the user cannot be a tie line user in a satellite without its own trunk line.

- The Follow-Me call forwarding feature is carried out until
  - the maximum permitted number of FWD destinations (set by AMO) is reached
  - a voice terminal is reached via FWD, where the FWD feature is not activated
  - the call cannot be switched
- Hunting group master.: In a hunting group, call forwarding is only possible from the master telephone. Call forwarding by other members of the hunting group is ignored in the Follow-Me sequence.
- Pilot hunting group: FWD is only possible from the last telephone in the sequence.
- CHESE functions are not counted as FWD steps in the sequence. However, call forwarding is counted if activated as such, either at the secretary’s or the executive’s telephone (even if call forwarding from EXEC to SECR is activated).
- Calls forwarded to exchange or tie-trunks are routed to the attendant, if the forwarding user does not have the appropriate authorisation.
- The call station remains available for outgoing calls and other functions while call forwarding is activated.
- If a caller wishes to initiate camp-on signalling or call override, this is not attempted until the last destination in the Follow-Me sequence is reached. Camp-on or override protection must be set at the Follow-Me destination, if required, i.e. these settings function regardless of the settings activated at the first telephone in the Follow-Me sequence.
- A maximum of all station users can have call forwarding activated at the same time.
- Device call forwarding. Device call forwarding which can be activated for non-voice terminals in addition to normal call forwarding cannot be used throughout the network.

5.2.7 Delayed Call Forwarding on Busy (DCFOB)

Delayed Call Forwarding - On Busy gives the called Keyset/DFT the ability to handle a second incoming call on the same line before the Keyset's Call Forwarding - Busy takes effect upon expiration of a timing period. Calls from stations receive special audible ring tone (ring/beep repetitively) as an indication that the called Keyset/DFT is busy and the caller can activate Callback Queuing on Busy. Calls from the public network receive normal ringback.

This Feature is based on

– Automatic Camp On - Second Call to offer the call to the destination while CFB is delayed.

– Call Forward No Reply (CFNR) as forwarding mechanism after time out if the offert call was not accepted by the destination.

– Call Forward Busy (CFB) to define the destination, where this call will be forwarded to.

The called Keyset/digital system- or IP-telephone receives Camp-On indication (Call Waiting indication).

When Delayed Call Forwarding - On Busy has been set to be active on a Keyset/DFT, it functions whenever the System or Station Call Forwarding is active on the Keyset/DFT and includes Call Forwarding on Busy.

This includes Call Forwarding Station (Forwarding - Variable) types:

- Call Forwarding - Busy -

- Call Forwarding - Busy/No Answer -

- Call Forwarding - Busy - External an internal

If a call attempts to terminate to a destination which has a camp-on Delayed Call Forwarding - On Busy configured and a call is already camped on; immediate forwarding of the calling party based on the call forward busy destination takes place.

Restrictions

- This feature is not applicable for analog telephones, functional devices, attendants, DECT-cordless-telephones or telephones without display.
- The function can only be activates by a configurable option. There are no feature keys and no feature codes
- Priority of Displays
  Upon deactivation or timeout of other displays, the displaying Delayed Call Forwarding On Busy Information is redisplayed.
- Display During Dialing
  Delayed Call Forwarding On Busy Information will is not given during dialing.
- Elapsed Time Display (ETD)
  DCFOB displays temporarily overwrite ETD displays.
- SID/ANI information, if available, is displayed prior to forwarding the DCFOB (camp-on) call.

- Charge Display (IM specific): DCFOB displays temporarily overwrites Charge displays.
- Consultation/Flash (Consultation Hold)
  If a user with DCFOB is in the Call Hold or Manual Hold state with a local call or is being consultation held or is the originator of a consultation hold call; and a call attempt is made to their busy extension, the DCFOB feature is ignored and forwarding is honored.
- Connect Key/Delayed Call Forwarding On Busy. When the PICKUP key LED or CONNECT key LED are flashing, a Camped-On call is pending. Either the PICKUP or CONNECT key will allow the user to pickup the call.
- Call Forward - All (unconditional) will override (supersede) DCF-OB
- Pilot Number Access
  A hunted call to a busy destination does not honor Delayed Call Forwarding On Busy and camp-on to the busy destination. Call forwarding is not executed unless the hunt group option indicates call forwarding.

5.2.8 Network-wide CFNR after Transfer

Call forwarding no reply for transferred calls functions exactly the same as for direct calls. Here it is irrelevant whether the call forwarding destination is in the same system or in a network.

The call forwarding destination can be:

- Any terminal unit in the network,
- A PSE (paging system),
- An attendant console,
- A user in the public network,

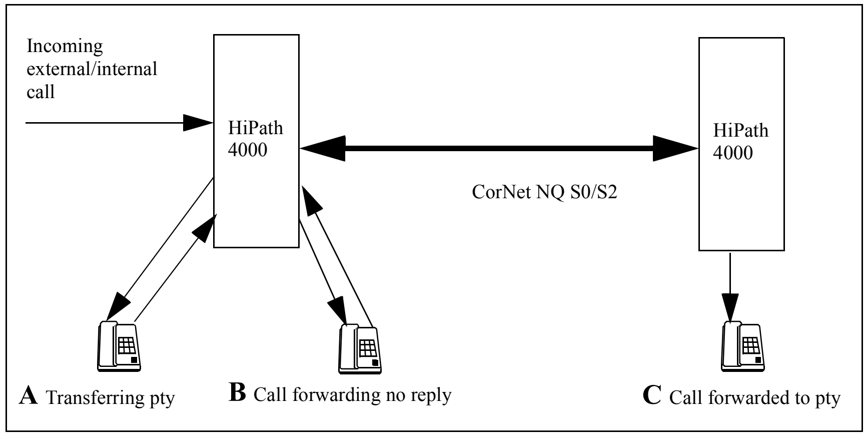
Example

An incoming call to **A** (user or attendant console) is transferred to user **B** whilst in the calling state.

User **B** has set up call forwarding to user **C**.

With expiration of the call forwarding timer, the call is now forwarded to user **C**.

A recall is received by **A**.



5.3 Call Forwarding-Extended (CFW-E)

The CFW-E (TC-AUL) feature represents a special call forwarding variant. With CFW-E, the CFW-E subscriber (B-party) can determine the call forwarding destination on busy/no answer in a series of call diversions comprising any number of stations. This functionality enables call diversion on busy/no answer to be executed to a specified destination (B-fix) as an alternative destination after the call has been forwarded multiple times.

Special Considerations of Call Forwarding - Extended:

- The CFW-E subscriber (B) must be the primary called party. The feature does not work from CFW-E subscribers that were reached as a result of call diversion.
- The feature must have been enabled centrally on the system.
- The CFW-E subscriber must have a subscriber-specific class of service.
- The call forwarding feature must have been enabled centrally on the system.
- The call forward on busy and call forward on no answer features do not have to be enabled at any of the stations affected in order for the feature to work.
- It is not necessary for the last station (D) in the diversion series to have a call forward on no answer/busy destination (D-fix) programmed for it. A diversion destination only has to be programmed for station B (B-fix).
- The CFW-E subscriber has to be a digital system- or IP-telephone. With some device types this feature is not operable (see listing above).
- The calling party as well as the stations in the CFW series and the CFW-E destination can be any device type.

- The protocol "ECMAV2" or "PSS1V2" has to be selected for networking functionality with this feature between systems.
- The CFW-E no answer timer setting used is the one set for the last station in the CFW series.
- The feature only supports voice calls.
- The number of call forward steps defined in the system data does not apply for the CFW-E feature.
- If the destination B-fix is to a party E who has also activated a CFW destination, this will not be followed.

Restrictions

- If a call forwarding chain involves the public network the CFW-E feature is not operable.
- If station is an attendant console, the CFW-E feature will only be executed on busy.
- As call diversion on no answer is not possible for attendant consoles, the CFW-E feature will not be working in this situation.
- CFW-E does not work if the CFW-E subscriber is part of a hunt group
- If the call diversion series involves a hunt group, the CFW-E feature is not executed.
- If the destination in a call diversion is a ACD-Application, the CFW-E feature is not executed.

5.4 Call Forwarding - No Answer

If a user who has activated call forwarding - no answer at his station (i.e. preset call forwarding destination must be programmed) does not answer a call within 20 seconds, the call is automatically transferred to the destination station entered.

- Call forwarding - no answer can be activated
  - to subscribers throughout the network
  - to subscribers in the public network and to subscribers in PABXs behind the public network under the following conditions:
    - for voice connections
    - with special classes of service
    - Charges for calls from the communication server to the public exchange must be borne by the forwarding party.

- Call forwarding - no answer is activated when a destination has been entered for preset call forwarding but no call forwarding function is activated.
  - If preset call forwarding is not activated, call forwarding - no answer is automatically activated.
  - The features "call forwarding - no answer" and "preset call forwarding" can be used in the system independently of one another, but have the same programmed destination.
- Prevention of call forwarding - no answer
  - when the connection between the caller and the called station is not permitted (station-to-station restriction). This restriction is optional and can be activated for each system using the administration and maintenance system.
  - The call remains at the initially called station. The caller does not receive a notification.
- Call forwarding - no answer is ignored (call remains at the initial called station) in case of
  - a call is from the destination station,
  - callback,
  - recall because the called station went on-hook in consultation status,
  - a DSS call,
  - ian timed reminder call,
  - ihe destination station is busy and has not activated console answering priorities or "call waiting - terminating",
  - calls forwarded via fixed call forwarding.
- After the transfer, the initially called station is then once again available for incoming and outgoing traffic.
- After each call forwarding - no answer, the next call will first be signalled at the initially called station again.
- The call goes to the destination station only; if call forwarding - all calls, call forwarding - no answer or a hunting group is activated there, it is ignored.
- Call forwarding - no answer applies also to DID calls.
- It is possible for several stations to have a common destination station for call forwarding - no answer.
- If call forwarding - all calls is activated at the initially called station, call forwarding - no answer is deactivated for as long as call forwarding - all calls is in effect.

- If personal call forwarding - no answer is activated for a hunting group station, this call forwarding - no answer is ignored when station hunting is carried out (hunting group call forwarding - no answer function is independent of station user call forwarding - no answer).
- All station users are authorised for call forwarding - no answer.
- A maximum of all users can have call forwarding - no answer activated at the same time.
- In the event of "callback on no answer", the callback request is entered at the initially called station or the destination station, depending on which station is being used when the entry is made.
- If a station user wishes in general to prevent call forwarding - no answer in the combination preset call forwarding - all calls/call forwarding - no answer, this user can be assigned the appropriate class-of-service by the administration and maintenance system.
- The administration and maintenance system can be used to define the timeout for call forwarding - no answer (default value is 20 seconds).
- The administration and maintenance system can be used to restrict call forwarding - no answer to outside calls.
- Calls forwarded to a busy user by the call forwarding - no answer function are signalled as waiting calls.

5.5 Call Pick up

5.5.1 Call Pickup (group)

A station user in a call pickup group can answer a call to another station in his group from his own telephone set.

1. A call pickup group may comprise analog and digital system- or IP-telephones. A call to any station in the group can be picked up at any other station in the group.
2. The user groups for the features "call pickup" and "call park" are identical.
3. All station users can pick up calls if they belong to the pickup group.
4. A station user can belong to one pickup group only. A group formed for an integrated executive/secretary system (executive/secretary, first secretary/second secretary) does not count as a call pickup group.
5. Signalling on digital system- or IP-telephones

- When a call is made to a member of a call pickup group, the called party is displayed on all other digital system- or IP-telephones in the group. The display of the called party's Digite shows the available calling line information.
- If the call is not answered after 3 rings, an alerting tone is sent to all digital system- or IP-telephones in the group.
- The alerting tone time-out can be set by means of administration and maintenance for digital terminals:
  – single ringing tone, only when the terminal is in the idle condition;
  – single ringing tone when the terminal is in any condition;
  – triple ringing tone when the terminal is in any condition;
  – no alerting tone.
- The display and alerting tone are only given if the display is not already being used for other procedures (e.g. dialling, programming, call signalling).
- If a call is being signalled in the group and, before it is answered, a direct call arrives at the telephone set in question, the display changes over to signalling the direct call with normal ringing.
- If some other display blocks the display panel for a waiting call pickup signal, the station user can "release" the display for the signal by ending his activity or, if a direct call is being signalled, by answering it.
- In the event of two or more simultaneous calls in the group, the calls are signalled in the order they are received. (The next call is not signalled until the first has been answered).
- Calls initiated from called station users are signalled at their telephones only.

6. During a consultation call, the station user initiating the call can press the TR key to return to the station user on hold, even if a second call is being signalled.
7. If there is a second connection (without signalling) in the call queue during signalling of the second call, the second call has priority after the call pickup key or TR key has been pressed.
8. If a secondary call (effective on digital system- or IP-telephones) is not answered, the call can be answered by all members of the same call pickup group like a first call.
   - Call number and name of the called subscriber are displayed alternately on the digital system- or IP-telephones of the same call pickup group, the call pickup group key flickers fast, and the call is acoustically signalled by means of a delayed alerting tone.

- Additionally, secondary calls can be picked up by members of the own call pickup group in the following situations:
  - Secondary caller is in consultation condition
  - Secondary caller is in secondary consultation condition
  - Secondary call is a recall
  - Secondary call is a recall in secondary consultation hold
  - Secondary call has been transferred by going on-hook or by pressing the transfer key
- The following applies to call pickup groups made up of executive/ secretary facility:
  - If the executive forms a team together with other subscribers external to the executive/secretary facility, and if the executive has not activated secretarial intercept, the specified features can be used.
  - If the secretary forms a team together with other subscribers external to the executive/secretary facility, secondary calls to the busy secretary initiated by other subscribers of the same call pickup group cannot be picked up.
  - If the secretary forms a team together with other subscribers external to the executive/secretary facility, secondary calls to the busy secretary can be picked up by other subscribers in the same call pickup group.
  - If the secretary forms a team together with other subscribers external to the executive/secretary facility, secondary calls intended for the executive can only be picked up by the secretary. If the secretary is busy, secondary calls **cannot** be pikked up by subscribers in the same call pickup group.

9. Call pickup without lifting the handset (digital system- or IP-telephones)
   - On a telephone with handsfree equipment, the handsfree equipment is activated by pressing the flashing call pickup key in the idle state; the call can be answered in this way.
   - On telephones without handsfree equipment, pressing the call pickup key in idle state is ignored.
10. On the digital system- or IP-telephones a call can also be picked up in the call state
    - during a first call,
    - in consultation or alternating state.

- The first call is automatically put on hold after the call pickup key is pressed.

11. Answering in the event of two or more simultaneous calls in the group: An answering station user automatically picks up the call which has been waiting the longest (answer in the order of arrival). The other calls are retained.
12. A caller with digital system- or IP-telephones/an attendant is shown which subscriber he has reached after the call has been picked up.
13. A call can be further serviced after it has been picked up.
14. A station user on hold as the result of a consultation call or call pickup cannot initiate a consultation call; a station user, however, can pick up a call in this state.
15. Station users in call pickup groups may also use their direct station select keys (DSS keys) for call pickup within the same group.

5.5.2 Directed call pick-up

Directed Call Pick-Up enables users to pick up calls outside the call pick-up group.

The user dials a feature access code followed by the desired extension number (which is different than the usual feature access code for network-wide call pick-up). With this feature, any extension (e.g. between different groups on the same switch and also network-wide) can pick up any other extension.

The feature itself is controlled by means of a class-of-service parameter for authorisation of the facility.

Feature Activation/Deactivation

The Directed Call Pick-up can be activated for calls in one of three states:

- Ringing
- Camped on
- Keyset manual hold.

By dialling the Directed Call Pick-Up facility a station user can answer a ringing extension locally and network-wide within the HiPath 4000 network or retrieve a call which is camped on or manually held at another extension, locally and network-wide within the HiPath network.

The feature access code can be programmed (entered) by the user under a NAME/DDS (direct destination select) key. To use the feature in conjunction with the NAME key, the user presses the key and suffix dials the extension number to be picked up.The key or the feature access code for Pick-up Group cannot be simultaneously used to invoke directed pick-up. The keys and the feature access codes of these two features are different.

There is one pick-up key which is used for group call pickup and directed call pickup. One PICK with the following digits of destination dialing is used for direct call pickup and pressing the pick-up key twice (PICK-PICK) is used for group pick-up and pick-up of campon call.

- Any authorised extension is able to pick-up any other extension (except AC) whether it is:
  - in the same pick-up group
  - in a different pick-up group
  - in no pick-up group
  - on the same Communication Server
  - network-wide
- It is not possible to pick-up a call in another group, using the access code for another group.
- It is not possible with this feature to pick-up a call that has been parked using the existing HiPath 4000 group park feature.
- In a CHESE configuration, calls for the executive can be picked up by dialling the number of the secretary's extension.
- A directed call pick up cannot be made from members of a conference call. The master of the conference call can still bring in other members using pick-up as long as the conference has not reached the maximum number of members.
- The feature is controlled by means of class of service (COS) parameter and can also be blocked in any one node (picking or called) in the network.
- The AC is able to pick-up calls at other extensions. However, other extensions are not allowed to pick-up calls at the AC.
- E-DSS1 and cordless devices can invoke the feature

5.5.3 Network-wide Pick-Up Group and Network-wide Call Park

Network-wide Pick-up Group is an enhancement to the Group Pick-up feature.

The enhanced Pick-up Group provides the capability for members of the pick-up group to reside on different nodes in a HiPath 4000 CorNet NQ (or heterogeneous DPNSS) network. In addition, the limit governing the number of devices, regardless of type, specified in a group is lifted to 255 devices.

The limit of number of groups per node is by default in configuration 255 groups.

The signalling level is automatically changed when devices are added to a group if a set threshold level is exceeded. The threshold levels and the notification changes for digital system- or IP-telephones are as follows:

- local devices <= 25 : full-signalling

  (Initially the display and tone information will be deactivated, leaving only LED indication within the group. This occurs when the number of digital devices in the group exceeds 25).

- local devices <= 75 : Only LED signalling (no display or tone)

  Display signalling is always determined by the local settings in each node. Accoustic signalling (periodic alerting and group call ring type) is always identical throughout the network, determined by the node where the called party resides (e.g. the "Master node" for the signalled call).

- local devices > 75: No signalling.

  The LED indication will also be deactivated when a further threshold level of 75 is exceeded. Parallel ringing (Group call) is deactivated when the signalling level is set to none.

Even if the setup is "No Signalling", a user may benefit from the feature "Display on Demand" which enables the total signalling options to be requested via a feature access code. This is deactivated if the user goes ON-hook .

The activation / deactivation of the display / tone notification and the LED indication is administerable, and is activated when the local group population reaches the respective threshold limits.

The Group Park feature is also enhanced to include all group members, whether local or on remote nodes, i.e. the user groups for ’call pick-up’ and ’call park’ are identical, as with previous versions.

Members of Team and executive/secretary configuration can also be members of a network wide pick-up group (regarding restriction of the specific configuration).

Unlike Group Pick-up indication, the Group Park feature has always a LED indication which cannot be switched off (unless one does not have a programmed park key at all) . The LED indication for the park feature is independent from the size of the pick-up group.

Feature Operation and User Interface

The local pick-up groups in different nodes are connected via networking signalling in order to provide network-wide call pick-up groups. Each node knows only about partner groups in other nodes (remote links) , but not about individual group members in other nodes. There two ways to provide information about these individual remote group members:

- Broadcast signalling :
  provides information about the member who is picked up. Not possible for group members from nodes other than HiPath 4000.
- Pick-up execution :
  establishes the connection between the calling party and the picking party. The messages used for "pick-up execution" do not depend on whether broadcast signalling has been provided or not. If no calls are available in the pick-up queue, consecutive pick-up attempts are made to all remote links which do not provide broadcast signalling.

The Network-wide Group Pick-up feature helps to inform all remote links whenever a local group member gets an incoming call which is available to be picked up, and also informs all remote links whenever such a call is no longer available for pick-up (e.g. it has been answered, or the caller hangs up).

The oldest ringing call is picked up. The exception to this is when a node (e.g. in a hererogeneous network DPNSS1) does not send network signalling . In this instance any call in the local queue is picked up before the none signalling remote nodes are interrogated (pick-up execution).

The feature can also be used to park a call to a Group Park position. In principle only 1 park slot per group is available.

Activation / Deactivation of the feature does not apply from the user terminal.

All Terminal device types that are currently able to invoke the existing Local Group Pick-up and Local Group Park features are able to use the Network-wide features (note, E-DSS1 terminals can invoke pick-up but cannot invoke park, AC terminals cannot invoke the features).

The User Interface for the different devices are affected as follows:

Call Pickup Feature Invocation from Analog telephones

- User hears terminal(s) in the group ringing.
- User goes off-hook and keys the feature access code.
- User is connected to the calling party of the oldest ringing call.
- The rung telephone stops ringing and returns to the idle state.

Call Pickup Feature invocation from Digital system- or IP-telephones

There are twoways of invoking the Group Pick-up feature :

- Group Pick-up key (when configured)
- Feature Access Code

Exception situations

- There could be a situation (when broadcast signalling is deactivated) that a user hears an instrument alerting nearby and requests group pick-up, only to be connected to a different instrument, since this was the longest alerting call in the queue. If a user attempts to pick-up an alerting call, but for whatever reason the call is no longer available for pick-up, then the user will be connected to the next alerting call in the queue, which will be different to the call displayed.
- The exception to this is when the alerting call is on a different node to the picking party and is no longer available for pick-up. In this instance the picking user will get the message 'Not Possible'.
- A user in a network-wide pick-up group, even if a subscriber of a node with broadcast signalling , has no display indication when picking up calls from the heterogeneous nodes.
- A user has no display indication in case of no broadcast signalling (e.g. when a call from a heterogeneous network node is picked up).
- If more than one call is parked simultaneously, a user picking up a parked call is always connected to the locally parked call.
- The Group Park LED may remain active, showing that a parked call has been queued,e.g. simultaneous parking.

5.5.4 Network-wide team functionality

The feature "network-wide team functionality" is a combination of existing features with specific modifications, e.g., to adjust the user interface to the one of CHESE. The used basic features are:

- Keyset
- Network-wide call pickup
- Several options for call forwarding

Prerequisites for the installation and usage of the "network-wide team" are:

- Networking of the systems with CorNet NQ
- HiPath 4000 Manager
- Closed numbering plan
- All network-wide signalling relationships are created by the feature "network-wide call pickup groups".

Executive/Secretary systems as a "network-wide team"

Configurations that are created as a "network-wide team" have the same basic functions as CHESE. These are:

- Executive line
- Private line
- Secretary line
- Network-wide direct calling
- Network-wide signalling of the called person
- Direct call and response
- Deactivation of secretarial intercept
- Deputy circuit
- Conference corner telephones
- Ring tripping

The remaining features, such as callback, camp-on, redialing, second call, voice server with the possibility to leave a message on the mailbox of the executive or the secretary, etc. can also be used by every member of the team.

The members of a "network-wide team" can be situated at different nodes of the network. The team members can use all features listed above.

Executive line

Every executive has an executive line (executive phone number) to which direct calls and calls forwarded by the secretary are directed.

The secretary or second secretary also has access to the executive line of the respective executive. Calls to the executive are signalled both visually by a fast-blinking LED and audibly by an alerting tone at the secretary's. The executive only gets a visual signal by a fast-blinking LED. When either secretary or executive answers the phone, the LED is glowing.

The secretary can have - by a change of line - a different incoming and outgoing call on each line. The current call is held automatically, showed by a slow-blinking LED.

Private line

Every executive can have a private line (private telephone number). Incoming and outgoing calls on this line can be done independently of the executive line. There is no signalling for these calls at the secretary's. The private line can be forwarded independently of the executive line. The call forwarding is signalled neither at the display nor by an LED.

Secretary line

Every secretary has an private line (phone number) on which her "personal" calls come in. Usually, "personal" calls of the secretary are not signalled at the executive's (private line of the secretary).

Network-wide direct station

The executive has established a direct station selection key for every secretary, by which he can reach her. Of course, a secretary can also have a direct station selection key for each of the executives. As with CHESE, the direct station selection calls cannot be forwarded. An active call forwarding at the direct station destination will not be carried out.

Network-wide signalling of the called person

The LED of the direct station selection key shows the free/busy signalling of the called person. The following states are shown:

– LED off	Line is free
– LED on	line is busy
– LED blinks	Person is being called. the call can be taken over by pressing the station selection key

Direct call and answer system

Because of data protection regulations direct call and answer systems (intercom systems) can be installed only by the administration and only on especially marked direct station selection keys. Direct call and answer systems are also possible on a network-wide basis.

Deactivation of secretarial intercept

Every secretary can forward calls for the executive to her telephone by pressing the key for deactivation of secretarial intercept. Incoming calls for the executive are signalled to her audibly and optically by display and blinking LED of the called executive line. An alerting tone and a notice at the display of the digital system- or IP-telephoneis applied to all other secretaries'.

If another secretary presses the key for the "deactivation of secretarial intercept" device, she then will be the "active secretary". Calls for the executive are then forwarded to her phone. (Remote call forwarding). The executive's display in idle condition always shows the name or the telephone number of the secretary who is his "active" secretary at the moment.

Every executive can also forward calls to his secretary by pressing the key for deactivation of secretarial intercept.

LED for secretarial intercept off	call forwarding to the first secretary's is not active; the first secretary does not get calls for the executive
LED for secretarial intercept o	call forwarding to the first secretary's is active; she gets the calls for the executive

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IMPORTANT: This feature is essentially different from CHESE!

Deputy circuit

Activation of deputy circuit

The "active" secretary can forward all calls for the executive to a secretary within the same team (who is pre-set by means of the administration and maintenance system). Then, the LEDs of the keys for deactivation of secretarial intercept are switched off at the secretary's forwarding the calls. Now, the LEDs of the keys for deactivation of secretarial intercept appear at the deputy secretary's. The LED of the deputy key does not glow at her phone. Calls for the secretary are also forwarded to the deputy secretary's.

Important: The destination of the deputy circuit cannot be programmed by the secretary; it is installed by means of the administration and maintenance system. Only a user within the team can function as destination of deputy circuit.

Deactivation of deputy circuit

A revocation of the deputy circuit can be applied by pressing the key for deactivation of secretarial intercept (not the deputy key) at the secretary's who earlier activated the deputy circuit. Thus, she deactivates the deputy circuit for the respective executive and is the "active" secretary for the executive again.

The call forwarding for calls for the secretary (personal calls for the secretary) is maintained. It can be deactivated by pressing the call forwarding key or by options on the service menu.

Conference corner telephones

Executive and secretary are always integrated into one call pickup group. A telephone assigned to the executive can also be implemented as a conference corner telephone into the call pickup group.

Ring tripping

A key for audible ring tripping can be installed at the executive's. Then, incoming calls are only signalled visually on the display and by the LED.

Network-wide team in combination with further features

Even though the network-wide team can be implemented as an executive/ secretary system alike CHESE, features are handled differently. The most important features are elaborated on in the following chapters.

Announcement system server

An executive call forwarded to the secretary that is not answered in an appropriate time is re-forwarded to another call forward destination. At the network-wide team, alike CHESE, one can choose between call forwarding to the mailbox of the executive or the secretary (department mailbox).The same is true (if provided) for calls for the executive forwarded to a secretary whose telephone is busy.

Callback

- In case of busy executive line

  A callback for calls for the executive forwarded to the secretary is always done as soon as the secretary's phone is free. The callback is carried out independently of the executive's phone being busy or free.

- In case of free executive line

  A callback for an executive with free line which is signalled at his secretary's is not stored in the executive's mailbox. In this case, a callback is carried out always when the executive line or the secretary line is free again.

Comparison of functions of "Network-wide team" and CHESE

Network-wide team	Local CHESE
Max. 4 secretaries and max. 4 executives. Every executive has one "active" secretary.	Max. 2 secretaries and max. 4 executives. Secretary is defined as first or second secretary.
Every team member can be in a different telephone system.	All in the same system.
The executive has one line key for private line and one for business line. He can pick up calls with either key and can place calls on either line. A call pickup key can be installed for picking up second calls.	The executive has one call pickup key to pick up second calls and calls forwarded to the secretary.
Private number is installed as a phantom line. It can be forwarded and answered directly.	Private number only adjustable via code number collective line. Separate call forwarding of the private number is not possible.
Executive has a key for the "deactivation of secretarial intercept" device. It glows when calls for the executive are forwarded to the secretary.	Executive has a key for the "deactivation of secretarial intercept" device. It glows when calls reach the executive directly; i.e. no call forwarding.
Executive has one direct call and answer key to the secretary. It glows when one of the secretary’s lines is busy. The LED blinks when the secretary is called.	Executive has one direct call and answer key per secretary. It glows when the secretary line is busy and blinks when the secretary is being called audibly.
Secretary has a line key for calls for the secretary’s number. It blinks when calls come in and glows when the line is busy.	No special key. No differentiation between secretary line and executive line.

Network-wide team	Local CHESE
Secretary has a line key for calls for the executive forwarded to her. It blinks when calls come in and glows when calls are answered.	Secretary has a call pickup key for each executive. LED blinks when secretary line is busy and the call is not signalled audibly. LED does not glow while call is answered.
An audible signal is applied to the "active" secretary’s for the respectively "oldest" call.	An audible signal is applied to the secretary’s phone for the respectively "oldest" call.
Secretary has one key for the "deactivation of secretarial intercept" device per executive. It glows when calls for the executive are forwarded to this secretary. State of the"deactivation of secretarial intercept" device is additionally signalled only at the executive’s. (No defined second secretary)	Secretary has one key for the "deactivation of secretarial intercept" device per executive. It does not glow when calls for the executive are forwarded to this secretary. State of the "deactivation of secretarial intercept" device is additionally signalled at the executive’s and the second secretary’s.
Provided another secretary presses the key for the "deactivation of secretarial intercept" device (not glowing at the moment), the call forwarding at the executive’s is then re-programmed to this secretary, signalled on the idle display at the executive’s. The LED at the formerly "active" secretary turns off.	If the second secretary, first secretary or executive presses the key for the "deactivation of secretarial intercept" device, the state is changed to activated/ deactivated.
Secretary has a key for deputy circuit. Pressing this key all call forwarding of executives having this secretary as "active" secretary is re-programmed to the deputy circuit destination. With no other number for call forwarding the secretary’s number is also forwarded to this deputy circuit destination. The LED of the key for deputy circuit does not glow.	Secretary has a key for deputy circuit. After pressing this key all calls for the executive are forwarded to the deputy circuit destination. The LED of the key for deputy circuit glows.
Activating the deactivation of secretarial intercept at a secretary’s changes the call forwarding destination of calls for the executive. Thereby, a call forwarding programmed by the executive (e.g. to an external phone number) is overwritten	Activation of deactivation of secretarial intercept does not change the call forwarding destination of calls for the executive. A call forwarding programmed by the executive is preserved.
Forwarded calls for the executive have to be re-forwarded to the call forward destination after a certain time.	No re-forwarding of calls for the executive forwarded to the secretary’s.
When the executive line is busy, forwarded calls for the executive have to be re-forwarded to the destination of busy-line calls.	No re-forwarding to the destination of busy-line calls for forwarded calls at the secretary’s.
Callback to busy executive line (if forwarded to the secretary’s). Callback is carried out when executive line becomes free at the secretary’s, independently of executive phone being free or busy	Callback to busy executive line (if forwarded to the secretary’s). Callback is carried out only when executive phone and secretary phone are free.

Network-wide team	Local CHESE
Callback to free executive line (if forwarded to the secretary's) cannot be forwarded to a mailbox, because a phantom line never has a mailbox. Therefore, it is a callback when phone is free.	Callback to free executive line (in idle state) is forwarded to the mailbox of the executive's, even if the call was forwarded to the secretary's.
Call forwarding, call forwarding of busy-line calls, and re-forwarding of forwarded calls to voicemail. Each executive can choose between voicemail entries into the mailbox of the executive's or the secretary's.	Call forwarding to voicemail. Within each telephone system one can choose between voicemail entries into the mailbox of the executive's or the secretary's.

5.6 Park to system

A subscriber (station/Attendant/ACWin) can park (in a system park slot) a party the user is talking to. After parking the present call a station user or Attendant/ ACWin is free to make other calls.

The digital system- or IP-telephone may be configured with a key for storage and retrieval of parked calls (e.g., Repdial or Park to System key).

If the parked party releases, the park position becomes available for new calls.

The Park to System feature is possible only during a stable two-party talk state.

A station/Attendant/ACWin is not allowed to park a call if:

– the requested slot is occupied (station only).

– all park slots are occupied.

– the party to be parked is ringing.

– either originating or party to be parked is in consultation.

– either originating or party to be parked is in conference.

– if the party to be parked is an Attendant/ACWin.

For an Attendant/ACWin, if the requested park slot is occupied but there is another park slot available, the present call is parked in the next available park slot automatically.

Devices that can be parked are:

- Analog telephone
- Digital system Telephone
- IP- telephone
- CO Trunk

- CorNet Trunk connections
- Profiset (functional device - DSS1)

System contains 10 System Park Slots (0-9).

100% of the telephones have access to the Park to System feature.

Upon retrieval of a Parked Call the station user will not receive confirmation tone.

Restrictions and feature interaction

Display features

- Priority of Displays on Keyset / digital system- or IP-telephone : Displays treatment when initiating the Park to System feature overwrites existing displays .
- Time and Date Display : Displays treatment when initiating the Park to System feature temporarily overwrites existing displays .
- Displays During Calling : Displays to the destination, when initiating the Park to System feature, temporarily overwrites existing displays .
- Elapsed Time Display : When Party C retrieves a Party B (Trunk) call which was parked by Party A (Station), the display of Party C shows the Elapsed Time Display after 5 seconds (or time based on system administration of display timer).
- Repeat ID : When a call is placed on system park and is then retrieved by the same station or another station, the display generated by the Repeat ID key is unaffected (the same as before park) except that the identity of the line in use may be different.
- SID/ANI: Displays to the destination, when initiating the Park to System feature overwrites existing displays .
- DNIS: Displays to the destination, when initiating the Park to System feature overwrites existing displays .
- Preview Feature
  In addition to its normal functioning, use of the Park to System key while in preview mode deactivates preview mode.
  Preview cannot be used to preview a line on Call Park - System.
  A Keyset user that is Call Parked - System can use preview.
- Caller ID Blocking/Unblocking/Automatic Recall on Held Calls (Recall)
  CID-B/U status is honored whenever an automatic recall (e.g., automatic recall on held call or station transfer security) occurs.
- Charge Display (IM specific)
  Displays treatment when initiating the Park to System feature overwrites existing displays .

- Display Suppression: Displays for Park to System supercedes display suppression.
- Call Log (Optiset E): Displays treatment when initiating the Park to System feature temporarily overwrites existing displays on the Keysets/DFTs involved.

Holding features

- Consultation/Flash (Consultation Hold)/ (Park to System)
  - Parking a call from a consultation talk (3-way party) state is not possible.
  - A Park to System destination cannot initiate a consultation call.
  - A station user can Park to System a destination from "consultation dial" (from A-B) state.
- Automatic Recall on Held Calls
  - Upon recall of a parked call, all devices with the line appearance of the prime line of the device, which parked the call, are alerted.
  - Recall and intercept to the Attendant take place as per Automatic Recall on Held Calls options are set in the customer database.
- Hold Basic/Manual Hold
  - When a user has parked a call and leaves the call parked until the recall timer expires, the system will attempt to recall the parking user. If the user is non-idle, the system restarts the park recall timer. When the recall timer expires again, the recall is attempted again
  - When an user has parked a call, this user can be removed from system (via AMO) before the recall timer has expired. If the recall timer then expires, the parked call is routed to the next attendant.

Busy Line features

- Internal Call Queuing - Standby
  - If the party-to-be-parked (to system) has a pending call waiting, they can NOT be parked.
  - If the party initiating the park to system has a pending call waiting, they can initiate the park to system feature.
  - A parked (to system) analog telephone can NOT received a waiting call.
  - If a parked station (digital system- or IP-telephone) receives a waiting call and picks it up, the park slot is vacated automatically.

- When conversing with a party and upon answering a pending "camped on call", it is possible to system park the answered (picked up) call.
- Emergency Disconnect of a party Parked to System is possible.
- Second Call
  A party Parked to the System can receive a second call; the user must disconnect prior to answering the second call.

Station Pickup features

- Call Pickup - Group
  - It is possible to park the current call while a Group Call Pickup call is pending.
  - If the parked station receives a waiting call and picks it up, the park slot is vacated automatically.
  - Upon picking up a call via group call pickup, it is possible to system park the picked up call.
- Call Pickup - Directed

  Upon picking up a call via directed call pickup, it is possible to system park the picked up call.

Multiline Appearance

Parking a call is based on the device being used, not the line. Upon recall of a parked call all devices with the line appearance of the device that parked the call are alerted. (Lines marked for ringing do so on the respective devices).

Phantom Lines

- A station while active on a phantom line can park another party in the system. (Orig.)
- A station user when conversing with a party which is using a phantom line, can system park that party. (Term)

Intercom features

Dial Call (Device) (Speaker Call - One Way)
A Speaker Call - One-Way cannot be parked. If this interaction is attempted, the caller receives "Park to system" (line 1) and "Not possible" (line 2) on his/her display.

Com Group Speaker Call - Two-way
A COM Group Speaker Call - Two-way can be parked.

COM Group (Intercom - COM Group Call)
A station user active in a COM Group call can initiate the Park to System feature.

Voice Calling (Speaker Call - Two-Way)
A Speaker Call - Two-way can be parked.

Speaker Call - One-way Broadcast
A Speaker Call - One-Way Broadcast cannot be parked. If this interaction is attempted, the caller receives "Park to system" (line 1) and "Not possible" (line 2) on his/her display.

A station user active in a speaker call after establishing a two-way connection can initiate the Park to System feature.

Integrated Executive / Secretary features

Telephones configured into an Integrated Executive Secretary System have access to this feature. A Private line has access to this feature.

Other features and functions

- Park to System can be initiated while active in handfree Operation.
- A bridged connection can not be Parked to system.
- The Park to System key can not be configured by the station user .
- If a station user system parks a call from station A and the then relocates station A, via Autoset relocate to station B, the system park is maintained.
- CDR/SMDR
  The call park duration is registered against the station (device) that parked the call.
  Upon retrieving a call from a park slot, CDR records against the retrieving party.
- Park to System can be accessed from this telephone when not active is baby listening (monitoring) mode.

Devices and Terminals

- An Analog Telephone user has access to the Park to System feature, and can be Parked to System.
- A Digital system- or IP-telephone user has access to the Park to System feature, and can be Parked to the System.
- A Keyset user has access to the Park to System feature, and can be Parked to the System.
- The Attendant has access to the Park to System feature but cannot be parked.
- A Profiset user does not have access to this feature but can be parked to system.

Networking

- A call via Primary Rate interface can be Parked to the System.
- A CorNet party that has been parked is a passive party and does not have access to activate other features and functions while parked.
- A CorNet party that has been parked can have calls camped on. The CorNet party must disconnect to service the call, which is waiting.
- A call via TIE Lines (analog and digital) can be Parked to the System.
- A DID (Direct Inward Dialing) call can be Parked to the System.
- A DIT (Direct Inward Termination) call can be Parked to the System.
- A call initiated via LCR can be Parked to the System.
- A Central Office (CO) (analog and digital) call can be Parked to the System.

5.7 Call hold (analog telephones)

Calll hold allows an analog telephone user to place a trunk or station connection on hold freeing up the line for other calls. Once a party is placed on hold; only the same station that placed that party on hold can, retrieve the party on hold. All analog telephones have the capability to place a party on hold.

A subscriber can hold the party the user is talking to in a private location, subject to these conditions:

– Only the station, which held the call, can retrieve it.

- While a call is on hold, the party that held the call is free to make other calls.
- Only one party can be held in the call hold location at a time. If a station attempts to put a second call on hold while another is on hold, the second call is placed on hold. The first call is removed from hold and reconnected to the station user activating hold.
- If the held party releases, the hold position becomes available for new calls.
- The same access code is used for the hold and hold retrieve functions.
- If a call is held and then retrieved, the user may place the call on hold again.

Devices that can be placed on hold:

- Analog telephones
- Digital system telephones
- IP- telephones
- Keysets
- Profisets (functional telephones - DSS1)
- TIE trunks (analog and digital)
- CO trunks (analog and digital
- CorNet connections containing the above devices and trunks

Call Hold is associated with an ITR group.

100% of the analog telephones have access to the Call Hold (analog) feature.

Feature Interaction and Restrictions

Short Announcements

The system can be configured to use short announcements. If configured the party placed on hold receives a short announcement that they are on hold.

Display features

Displays During Calling
Displays to the destination, when initiating the Call Hold (analog) feature, temporarily overwrites existing displays on the Keyset/Digital system- or IP-telephone

Repeat ID
A station can not initiate the Repeat ID feature while being held

SID/ANI/DNIS
Displays to the destination, when initiating the Call Hold (analog) feature overwrites existing displays on the Keyset/Digital system- or IP-telephone

Charge Display (IM specific)
Displays treatment when initiating the Call Hold (analog) feature overwrites existing displays on the Keysets/Digital system- or IP-telephones involved.

Holding features

Consultation/Flash (Consultation Hold)
A station can not retrieve a call from call hold when conversing with a consulted party.
A station being held (call hold) can not invoke the consultation feature.

Toggle
A station user toggling between two parties call can not place either party on call hold.

Automatic Recall on Held Calls

For all users, the system does not allow internal or external calls to remain on hold indefinitely. Calls will recall and alert the station that placed the call on hold, if the station that placed the call on hold is idle.

When an user has held a call and leaves the call held until the recall timer expires, the system will attempt to recall the holding party. If that party is non-idle, the system restarts the hold recall timer. When it expires again, the recall is attempted again.

When an user has held a call, this user can be removed from system (via AMO) before the recall timer has expired. If the recall timer then expires, the held call is routed to the next available attendant.

A call left on Call Hold is recalled to the party that placed the call on Call Hold after a time-out, or to the Attendant, dependent upon the system parameter (recall to station vs. Attendant) and the state of the party that held the call. Note that incoming public network trunk calls placed on Call Hold (analog), will recall to the original station, and then route to Attendant/Night Option Arrangement, if recalled analog station does not answer.

Hold Basic/Manual Hold / Exclusive Hold

An analog telephone user placed on manual hold by a Keyset can not initiate call hold.
A keyset user placed on call hold by an analog telephone user can not place the connection on Manual/Exclusice hold.

Park to Station / Group Park

An analog telephone user Parked to a Station can not initiate call hold. A user placed on call hold by an analog telephone user can not initiate Park to Station or Group Park.

Station Controlled Conference features

It is not possible to put a local station contolled conference on hold.

Only a network station controlled conference can be placed on hold by an user. In a network station controlled conference the holding Anate and the held conference must be in different switches. If the user is in consultation talk to another subscriber with a held station controlled conference and the user hook flashs, the consultation call will be released and the user will be reconnect to the station controlled conference. It is not possible to put the consultation call on call hold (analog).

Busy Line features

An analog telephone on hold can NOT receive a waiting call.

A Digital system- or IP-telephoneon hold may receive a waiting call.

If the party-to-be-held has a pending call waiting, they can NOT be held.

If the party initiating the call hold (analog) feature has a "camped-on" call pending, they can initiate the call hold (analog) feature. Upon answering a pending "camped-on" call, it is possible to call hold the answered (picked up) call.

Executive and emergency Override

Override of a party on call hold is not allowed.

A station user engaged in a three-way connection containing an overriding party cannot place the original or overriding parties on call hold.

Station Pickup

A station user placed on call hold cannot pickup a pending group and directed call pickup call.

Abbreviated dialing features
The access code for Call Hold (analog) can be configured into System Speed.
The access code for Call Hold (analog) can be configured into Station Speed.

Call Origination

A Bridged connection can not be placed on Call Hold.

Other features and functions

Automatic Set Relocate: Held calls will not be maintained upon relocation.

CDR/SMDR: The time the call was held is registered against the station that held the call. Upon retrieving a call CDR will record against the retrieving party.

Devices and terminals

A Profiset user does not have access to this feature.

Networking

A CorNet party that has been held is a passive party and does not have access to activate other features and functions while held. A CorNet party that has been held can have calls camped on. The CorNet party must disconnect to service the call, which is waiting.

5.8 Speed Dialling - System

1. Users and the attendant have access to speed dialling lists (memory in the call processing equipment).
2. The speed dialling lists contain internal and/or external call numbers which can be accessed by using the pushbutton set or feature keys; they are then sent automatically.
3. There is a limit of 22 digits per speed dialling destination (including the trunk group code for external destinations).

4. Maximum number of speed dialling lists per system: 16
5. Each user and each attendant has access to the system list (No. 0) and one of the 15 other lists.
6. Access to the system speed dialling lists is defined with the aid of administration and maintenance.
   - A group code is allocated for each of the 16 system speed dialling lists.
   - Each speed dialling list can be accessed by a group of users/attendants; in its maximum configuration a group may comprise all users/attendants.
7. The abbreviated call numbers in a speed dialling list are uniformly two or three digits long.
   The abbreviated call numbers (speed dialling numbers) may also be set at 4 or 5 digits, instead of the standard 2 or 3 digits. Regardless of the speed dialling number length, the number of possible speed dialling lists remains unchanged, although the contents of each list must be defined according to customer-specific requirements, depending on the function.
8. There are two ways of accessing the speed dialling lists:
   - Dialling one of the two codes for the one or two speed dialling lists per user/attendant and dialling the abbreviated call number. The two codes are the same for all users/attendants
   - Dialling a common code for system speed dialling and dialling the abbreviated call number. The code is the same for all users/attendants. The abbreviated call numbers in lists 1 to 15 must be greater than those in list 0.
9. Each speed dialling list may contain a maximum of 1000 destinations.
10. At most, 100% of users and attendants can use system speed dialling.
11. Management of the speed dialling memory
    - The speed dialling memory may be divided into lists of different sizes according to customer requirements.
    - The call numbers are entered in the speed dialling lists with the aid of administration and maintenance.
12. There is no toll/code restriction in force when speed dialling is used.
13. The System Speed Calling is local to a single node.
14. Manual suffix dialling for external calls. After the abbreviated call number, further digits may be dialled and these are sent out after the stored call number.
15. Automatic suffix dialling for external calls

- Automatic suffix dialling of the stored call number (maximum length 5 digits) of the attendant console of the PABX accessed through speed dialling (after timeout) if a user call number is not dialled.
- The timeout period is defined with the aid of administration and maintenance.

5.9 Speed Dialling - Individual

1. The individual speed dialling list contains internal and/or external call numbers which are accessed by using the pushbutton set and then sent automatically.
2. There is a limit of 22 digits per speed dialling destination (including the trunk group code for external destinations).
3. All dialling information entered on the pushbutton set (i.e. including * and #) can be stored.
4. Each user has access to an individual speed dialling list. The authorisation for an individual speed dialling list is allocated per station, via AMO.
5. Individual speed dialling lists are assigned to users with the aid of administration and maintenance.
6. 0, 10, 20, or 30 Speed Dial Individual entries will be assignable by Administration on a per station basis.
7. Memory for speed dial entries will only be allocated to a station when that station is assigned. .
8. Users can enter destinations themselves.
9. The maximum number of station speed dial groups (of 10 speed dial numbers) is based on the amount of memory allocated for this feature.
10. The Speed Dial - Individual is local to a single node.
11. The station speed index number, used for programming (storage) and dialing, will be 0-9 for telephones that are assigned 10 numbers.
12. The station speed index number, used for programming (storage) and dialing, will be 00-19 for telephones that are assigned 20 numbers and 00-29 for telephones that are assigned 30 numbers.
13. Toll/code restriction when individual speed dialling is used.

    If using individual speed dialling to set up a trunk call the user must have toll access.

14. Manual suffix dialling for external calls: After the abbreviated call number further digits may be dialled and these are sent out after the stored call number.

15. Entering a call number in the individual speed dialling list

    - A user can enter abbreviated call numbers with the appropriate unabbreviated call numbers in his individual speed dialling list.
    - Any number already stored under the abbreviated call number is overwritten.
    - Digital system- or IP-telephone: A number of destinations can be entered one after the other without having to start the procedure from the beginning each time.

16. Deleting a call number in the individual speed dialling list

    Any destinations entered by a user in the individual speed dialling list can be deleted

    – by overwriting with another destination

    – with the aid of the delete procedure.

17. Checking the contents of the memory locations in the individual speed dialling list (Digital system- or IP-telephone)

    - A user can check the unabbreviated call numbers stored under the abbreviated call numbers in his individual speed dialling list.
    - A number of destinations can be checked one after the other.

18. System/Station Speed Number - Chaining

It is possible to activate multiple features and functions (e.g., Call forwarding) in one Station or System Speed Dial sequence. Additionally, if multiple features require more digits then are available in a single Station Speed entry, allow the user to link one Station Speed Entry to another.

Allow a station user to configure a Station Speed Dial entry to link to another Station Speed Dial entry. Allow linking of up to 10 Station Speed Dial entries, in any order.

Example 1:

Station Speed Dial Entry 01 linked to Station Speed Dial Entry 11

(Note: The access code for Station Speed Dial access is "*55".)

– Entry 01: *3 (PIN access code) + 12345678901 (PIN) + *55 + 11 (next entry)

– Entry 11: 9 (LCR access code) + 15619231705 (Destination)

(Each entry 22 digits maximum)

Example 2:

– PIN access code + PIN + LCR access code + Destination Number

(Each entry 22 digits maximum)

19. This station speed feature is not applicable for attendants.

5.10 Callback

Each user can decide whether to allow ’callback on busy’ or ’callback on no answer’ for his or her call number, as required. This is intended to satisfy data protection requirements and particularly requirements of the civil services (public services) in Germany.

5.10.1 Callback on Busy

1. Callback is for voice calls only.
2. When a station is found to be busy, it is possible to enter a callback request, after which the connection is cleared down. As soon as the desired station is free, the callback is initiated to the requesting station . When the callback is answered, the desired station is automatically called.
3. A callback request is entered
   - for a internal extension
   - network-wide for a subscriber of the private network with CorNet NQ or Q-Sig (certification is necessary) if callback is provided in all nodes concerned
   - The "callback if busy" function can only be activated if the callback entry is enabled for the destination number dialled.
     The caller receives a negative acknowledgment in these cases:
     – analog telephone users receive a negative acknowledgment tone.
     – Digital system- or IP-telephone users receive a display output. This is "PROTECTED" in the case of HiPath 4000 users, and "NOT POSSIBLE" in the case of users in systems other than a HiPath 4000.
4. Maximum number of concurrent callback requests per A-station:
   - Analog and Digital system- or IP-telephone w/o display: 1 request
   - Digital system- or IP-telephone with display: 0 to 99 orders (standard: 5x callback)

5. The callback light is lit at station A as long as at least one callback or call queuing request is entered (Digital system- or IP-telephone).
6. Interrogation of stored callback requests on a Digital system- or IP-telephone
   - The entered requests can be displayed by means of scanning; the names, insofar as they are available, are also shown.
   - For Digital system- or IP-telephone, the memory contents are displayed cyclically, i.e. after the last callback request has been shown, the display begins again at the start.
7. Deleting a callback request
   - Manual deletion node-internal and network-wide.
   - Automatic deletion:
     – when station A answers the callback
     – if station A does not answer the callback within 20 seconds
     – if station A enters a new callback or call queuing request (analog telephone).
8. Station A is free for incoming and outgoing calls during the waiting time for the callback.
9. The callback waits until Station A and B is free
10. Answer the callback (station A answers)
    - Analog telephone: lift the handset
    - Digital system- or IP-telephone: lift the handset or press the loudspeaker key
    - After answering station A receives ring tone; station B is called
11. During the callback at station A, the now free station B is designated as busy for other incoming calls.
12. If station B uses his station line for an outgoing call during the callback process, the callback process is terminated and station A becomes free again. When station A answers it therefore receives internal dial tone. The callback is repeated later when the conditions for its execution have been met again.
13. If call forwarding - all calls, do-not-disturb, call forwarding - no answer or hunting group are activated for station A, these features are ignored for callback.
14. A callback to a busy station cannot be answered at another station using call pickup.

15. If station B has activated call forwarding
    - before the first call: the callback request is entered for the destination station of call forwarding.
    - after the first call: after the callback is answered the call is signalled at the call station.
16. All station users can use callback on busy.

5.10.2 Callback on No Answer

If a station user calls a station at which no one answers, he can enter a callback request and then clear down the call. The desired station user (station B) then initiates a callback to the station user who entered the callback request.

The requirement for the callback to take place is the next termination of a call by station B. Station B is called after the callback is answered by station A (analoges-, and Digital system- or IP-telephone without mailbox).

The requirement for the callback to take place is initiation during scanning in the mailbox (Digital system- or IP-telephone with mailbox).

If the callback implemented by selecting from the mailbox encounters a (now) busy user (user who entered the callback request), the call waiting tone is applied.

1. Entering a callback request
    - for a node-internal extension
    - network-wide for private network terminals with CorNet NQ or Q-Sig (certification is necessary!)
    - It can be administered via AMO whether 'Automatic callback – no answer' is to be allowed or prevented.
      Prevention of callback is signalled to the calling user as follows:
        – with an analog telephone through a negative acknowledgment tone,
        – Digital system- or IP-telephone users receive a display output. This is "PROTECTED" in the case of HiPath 4000 users, and "NOT POSSIBLE" in the case of users in systems other than a HiPath 4000.
    - The "callback on no answer" function can only be activated if the dialled station does not have a mailbox and if "callback protection" is not activated.

- For "callback on no answer" a station (Digital system- or IP-telephone) with mailbox can be configured to be treated as a station without a mailbox. In this case, the callback is not entered into the mailbox (paging for callbacks is not possible), but executed as soon as the station goes on-hook after completing a call.

2. Maximum number of callback requests which can be entered per A-station:
   - Analog- and Digital system- or IP-telephone w/o display: 1 request
   - Digital system- or IP-telephone with display: 0 to 99 requests (standard: 5 callback requests)
3. The callback light is lit at station A as long as there is still a callback request entered there (Digital system- or IP-telephone).
4. Interrogation of stored callback request on a Digite
   - The entered request can be displayed by means of scanning; the names, insofar as they are available, are also shown.
   - For Digite, the memory contents are displayed cyclically, i.e. after the last callback request has been shown, the display begins again at the start.
5. Deleting a callback request
   - manual deletion
   - automatic deletion
     – when station A answers the callback,
     – if station A does not answer the callback within 20 seconds,
     – if station A enters a new callback or call queuing request (Analog telephone),
     – at station B with mailbox, the callback request is deleted after the callback has been started.
6. Station A is free for incoming and outgoing calls during the waiting time for the callback.
7. The callback is delayed until the following criteria are fulfilled:
   - station B (Anate/Digite without mailbox) has terminated a call.
   - station B with mailbox has pressed the Start key during scanning in the mailbox.
   - station A is free.
8. Answering the callback (station A answers)
   - Analog telephone: by lifting the handset.

- Digital system- or IP-telephone: by lifting the handset or pressing the loudspeaker key.
- After answering, station A receives ring tone. Station B is called (station B has no mailbox).
- After answering, station A is connected with station B (station B has a mailbox).

9. During the callback to station A, the now free station B is designated as busy for other incoming calls.
10. If station B uses his station line for an outgoing call during the callback process, the callback process is terminated and station A becomes free again. Exception: station B has activated call waiting – terminating. If station A answers it therefore receives internal dial tone. The callback is repeated later when the conditions for its execution have been fulfilled again.
11. If station A has activated call forwarding - all calls, do-not-disturb, call forwarding - no answer or hunting group, these features are ignored for callback.
12. A callback to a station on no answer can be answered at another station by means of call pickup.
13. If station B has activated call forwarding - all calls
    - before the first call: the callback request is entered for the destination station of call forwarding.
    - after the first call: after the callback is answered the call is signalled at the call station.
14. If station B has activated call forwarding - no answer
    - before the first call: the callback request is entered for the station which is being called at the time of entry.
    - after the first call: after the callback is answered the call is signalled at the initially called station.
15. If station B is in a hunting group
    - entry of a callback request on no answer is not possible.
16. if station B has activated do-not-disturb
    - before the first call: the callback request is entered. The callback is carried out when station B terminates a connection after deactivating do-not-disturb.
    - after the first call: the callback request is entered. The callback is carried out when station B terminates a call either before or after the do-not-disturb state.

- If do-not-disturb is deactivated by means of a dial procedure (Analog- , Digital system- or IP-telephone without ANS key), the callback is carried out after termination of this connection.

17. All station users can use callback on no answer.

5.11 Call Transfer

In the case of Digital system- or IP-telephone the transfer function is carried out by means of a special key

With analog telephones , this function can be performed

- with the grounding button
- with the flash key, or
- by dialling a single-digit code.

Whether call transfer is to be performed via normal transfer or pickup is defined by means of administration and maintenance for the entire system. The procedure set in this way applies to all Analog telephones, except for the night answer extensions. Digital system- or IP-telephone can always perform normal transfer, irrespective of the AM setting. It can be said for network-wide application that the procedure which has been set in the node in which the active terminal is administrated is effective.

5.11.1 Transferring External Calls

1. When an external call is set up, the system checks that the user
   - has the necessary local or toll exchange access classmark for the dialled local area or toll call and
   - is allowed to dial that particular destination or route number.

   If such a user wishes to transfer a call to another user in the system, it depends on the system administration whether the recipient
   - requires the same classmark necessary for setting up the call
   - does not require the same classmark necessary for setting up the call

   If the transferring user is an attendant, no restrictions apply, i.e. an attendant may always transfer a call regardless of the recipient station's class of service.
2. In the consultation state the external call on hold can be extended with the aid of call transfer procedure to the consulted station terminal. The consulting user then no longer has a connection to the two other parties.

3. Transfer of incoming and outgoing external calls
   - transfer of tie trunk calls
   - transfer to an internal user
   - transfer to the attendant console
   - transfer of trunk calls

   Trunk calls can even be transferred to other trunk calls in some circumstances.However, it must be guaranteed that both circuits concerned must be cleared down when the call is terminated.

4. It is possible, with the aid of administration and maintenance, to define whether transfer is to take place through normal transfer or through pickup.

5. Normal transfer
   - The consulting user can initial call transfer before or after answering the consulted user.
   - On-hook transfer (Analog - /Digital system- or IP-telephone)
   - Transfer by pressing the UEG key (Digital system- or IP-telephone)
     – On transfer of an initial call the consulting user hears the dial tone.
     – On transfer of a second call (such as camp-on call), the transferring user is connected to the user with whom we was speaking before the second call was answered.
   - On transfer before a busy consulted user answers, the transferred call enters a retest state at the busy station.
   - If, after transfer, the consulted busy Digite user releases his initial call by pressing the TR key he hears dial tone and can set up a new call. The transferred call is not signalled until the consulted Digital system- or IP-telephone user goes on-hook.
   - If an exchange call is transferred to a busy station without the camp-on (knocking) protection or do-not-disturb features, or if these features have not been activated by the user, the camp-on (knocking) or call waiting signal is always applied if the recipient of the transfer is busy and if the recipient of the transfer is talking, i.e. in a state where the call waiting signal can be applied.
   - If the recipient does not answer the waiting call within a specified time (callback timer can be set between 20 and 40 seconds), then the transfer initiator is called back. If one of the above conditions is not fulfilled, the transfer initiator is called back immediately. An immediate callback is also carried out if the intended recipient has an analog station and is either participating in a consultation hold call, a three-way conference, or call toggling (flip-flop). If the call waiting signal is applied to the recipient's line,

the recipient may still set up a conference call, transfer a call, and release a call from conference hold state. The call waiting signalling is simply interrupted for the duration of these transactions.

- If the intended recipient has activated call forwarding - no answer, call transfer is rejected and automatic recall is executed.

6. Transfer through pickup
    - The consulted user picks up the call on hold by pressing the SIG key or UEG key or by pressing the dialog key to confirm the Service Menu option.
    - After the call has been picked up, the consulting user hears the busy tone.
    - If the user on hold releases his call before call transfer takes place the user picking up the call hears the busy tone.
7. Prevention of call transfer with automatic recall
    - on transfer to an unavailable station line,
    - on transfer to a call number which does not exist,
    - on transfer if all trunks are busy,
    - on transfer to a station terminal which is out of service,
    - on transfer to a station terminal which does not have the class-of-service corresponding to the call on hold,
    - on transfer to a user who is already waiting for a call,
    - on transfer of a trunk call to an external party,
    - on transfer through going on-hook during a consultation call to a user with the "do-not-disturb" feature activated (unless the consulting user is authorised to violate the "do-not-disturb" feature),
    - if the consulting user goes on-hook before call pickup,
    - if the user has not dialled or not dialled the full number in the consultation state,
    - upon transfer of private trunk calls set up with a specific code. If, however, a private trunk call is set up by a secretary within a CHESE configuration, the transfer prevention facility is overridden.
8. Call pickup at a station terminal which does not have the class-of-service corresponding to the call on hold is denied. The consultation call remains in force.

9. With the aid of administration and maintenance the transfer of outgoing trunk calls can be barred for all users of a system. As an exception, the night station can transfer (extend/assign) outgoing trunk calls during night service.
   - Reaction of the system to a transfer attempt
     - by going on-hook: automatic recall to the originator
     - with the UEG key: the keystroke is ignored (negative acknowledgment on the display)
   - Reaction of the system to a pickup attempt
     - with the signal key: the keystroke is ignored
     - with the UEG key: the keystroke is ignored (negative acknowledgment on the display)

5.11.2 Station User Transfer before answer over DSS1

An authorized user (not a functional terminal unit) who dials the complete number of an external user can now perform a consultation hold before the acknowledgement of the first connection.

This authorized user can then set up a consultation call to a user in the network or internally, and connect the two calls together when one of the parties called has answered.

"Toggle", return to the first call:

- If the consultation call has been answered (answer), the user performing the consultation can switch back to the first call using the hold toggle key, and also switch again to the consultation call.
- If the consultation call is not answered (no answer) and the user performing the consultation switches back to the first call using the hold toggle key, the consultation call is cleared.

Take note of the following:

- In the case of transfer (by going on-hook) before one of the parties has answered, the party performing the transfer is recalled and the consultation call cleared. If the transferring party does not answer, the first call is cleared and the transferring party released after a period of time.
- If the consultation call is in the talk state and the transfer is made by going on-hook, in cases of unsuccessful transfer (e.g. station C on the consultation call is not authorised/ station C is a functional terminal unit/ station C off-premises connected via analog tie line circuit) the transferring party is recalled and the consultation call cleared.

- If the first call is connected to a server, the consultation call has to wait for the acknowledgement of the first call in order to transfer the connection.
- A call transfer will not occur if the first call does not send an acknowledgment and the consultation call is connected to a server or central attendant console.
- If the consultation call is in the talk state and the external user does not answer, the user originally performing the transfer is not recalled.
- If the first call (external ISDN user) has answered and the consultation call not, the transferring user is recalled after a period of time (normal operation)
- If dialling for the first call is not completed through to the destination (not enough digits), then the dialling of the destination user can be completed when the party performing the consultation switches back to the first call and clears the consultation call in the process.
- The user performing the transfer receives no information on the progress of the first call held. A call transfer is only reliably performed if the user performing the consultation has answered.

Impact on other Features

- If the consultation is initiated very early, any LCR rerouting on the first call is deactivated. As a result, a possible traffic rerouting due to all trunks in the network being blocked is not performed for the first call.
- Call park cannot be used before answer.
- It is not possible to transfer an unanswered first call to an answered consultation call via an analog line.
- If a non registered external DSS1 connection is busy or rerouting takes place then no consultation or transfer is possible.
- Blind transfer by an Digital system- or IP-telephonel will disable display "not possible" in the event that the transfer does not take place (whereas, transfer via key results in this cas with the negating display).
- In some cases a call fowarding/CF no answer might be inhibited with this feature, e.g. when the executive is dialled, the call will go to the executive station directly and not the secretary in a CHESE configuration. By transfer to a hunt group, only the master number will be selected and no other hunt group members will receive the call.
- A consultation call over the same line cannot be executed -it takes place on a second B-channel.

5.11.3 Transferring Internal Calls

1. In the consultation state the internal call on hold can be extended with the aid of call transfer procedure to the consulted station terminal. The consulting user then no longer has a connection to the two other parties.
2. Transfer of incoming and outgoing internal calls
   - transfer of a call to a user
   - transfer of a call to the attendant console
   - transfer to an internal user
   - transfer to the attendant console
   - transfer to a user in the remote system via a tie trunk
   - transfer to an external party
3. With the aid of administration and maintenance the transfer of internal calls to external parties can be barred for all users of a system. By way of exception, the night station can transfer (extend/assign) internal calls to external parties during night service.

   Reaction of the system to a transfer attempt
   - by going on-hook: automatic recall to the originator
   - with the UEG key: the keystroke is ignored (negative acknowledgment on the display)
   - The same feature bit is used for extending and assigning.

5.12 Station Controlled Conference

A conference member has the capability to hold a conference, consult from conference, view conference members, create a conference upon pick up of a party, and add parties to a conference. The conference consultation/toggle feature is applicable to analog and Digital system- or IP-telephones (except functional and SIP- telephones).

The ability to pick from conference and add a party to conference or pickup a party from two party talk state and create a conference is applicable to digital telephones only (except functional devices).

A functional device (e.g., Profiset) is a passive conference member and therefore can not activate features and functions while in a conference. A functional device can create an Add-on (three-party) conference only; in this case, the other members are passive conference members and therefore can not activate features and functions while in an Add-on (three-party) conference.

1. This conference function does not require that a master control the conference. Each member in a Station Controlled Conference is able to operate independently. One or more parties can split from the conference at any time to initiate features, consult or add other parties to the conference.
2. There are eight possible conference positions available in a conference, per node.
3. Conference members maintain their positions in the conference when a member leaves a conference.
4. Of the 8 parties allowed in conference, 8 parties may be trunks.
5. An Attendant cannot be a member of a conference. An Attendant can not create a conference.
6. Conference and Consultation Conference Parties may consist of internal stations, TIE Trunks, CorNet Parties and non-CorNet trunks .

   The following can be a member of a conference (This party be referred to as conference member):

   – Analog telephone
   – Keyset
   – Digital system Telephone
   – IP-telephone
   – functional telephone
7. CO and Tie trunk is a purely passive conference participant. It has no way to add or release parties from a conference, and displays will not be given. A station connected to a TIE trunk may initiate a conference local to the node they are in.
8. The maximum number of conferences based on the number of parties in conference. There are 192 conference ports, each assigned as the station user enters a conference (e.g., 192/8 parties = 24 eight party conferences, 192/3 parties = 64 three party conferences).
9. One hundred percent of the analog and digital telephones (except functional devices) have access to the Station Controlled Conference features and functions.
10. Music is not given to the conference while the conference is on hold.

Add-On Conference (Three-party)

Add-on Conference provides the capability for a functional device (e.g., Profiset) to create a conference of up to three parties (total).

– A functional device can create an Add-On (3-party) conference.

– Functional devices as conference members are passive and do not have access to features and functions.

– Non-functional devices in an Add-On conference created by a functional device are passive members.

After the three-party conference has been set up, an override tone (3 sec duration) is sent to all stations.

On Digital system- or IP-telephone with displays, a visible signal is also provided: for the originator of the conference: "CONF. 1-2-3", for the other participants: "CONFERENCE".

If one of the participants hangs up, the remaining connection reverts to a normal, basic call. The display changes, indirectly signaling that the participant has really hung up and did not simply pretend to do so.

5.13 Enable/Disable Second Call Feature

Users with Digital system- or IP-telephone can enable/disable the second call feature for their terminal with a programmable key. A second incoming call is signalled acoustically and visually when a call is in progress. The first call is placed on hold when the second call is picked up.

5.14 Timed Reminder Service

1. Through entry of reminder data from his telephone set, the station user can arrange for timed reminder service, which consists of automatic calls at pre-specified times accompanied by a message from the system about the appointment.
2. The reminder data refers to the 24 hour period following input of the data (automatic daily repetition is not possible).
3. Maximum number of reminders which can be stored per station user;
   - For analog telephones: 1 reminder
   - For Digital system- or IP-telephone: this corresponds to the number of reminders which can be stored for each system.
4. Deleting a reminder
   - Two or more reminders can be deleted consecutively.
   - A new reminder can be input while the display <DELETED> is shown.

- If incorrect input is made during entry, all digits entered up until that point can be deleted and the correct reminder then entered.
- Automatic deletion following acknowledgment of the reminder call is possible.

5. Interrogation of reminders (on Digital system- or IP-telephone)
   - The station user can interrogate in chronological order the reminders which have been entered.
   - The chronological order is sorted by the system regardless of when the reminders were entered.
   - The time of day is entered with a 4-digit input (e.g. 0945 for the time of day 9:45 a.m.).
6. Reminder calls
   - On analog telephones; special call, One of the two special calls is used as the reminder call.
   - On Digital system- or IP-telephone with timed reminder key:
     – special call in idle state, one of the two special calls is used as the reminder call.
   - alerting tone in the call state
     – the display and the alerting tone occur in the call state only when the display is free.
   - On Digital system- or IP-telephone without timed reminder key;
     – special call in idle state, one of the two special calls is used as the reminder call.
   - alerting tone in call state
     – signalling in the call state if call waiting - is activated.
   - Duration of the reminder call. The maximum duration corresponds to the length of the timeout for recall.
   - If the station is busy at the entered time for the reminder, the reminder call is postponed until the station is free again (stations with call waiting - terminating are considered free).
   - If the reminder call is not acknowledged within the timeout duration, it is repeated once after 5 minutes and then deleted.
   - The reminder call can be answered during a call state. Any display which is shown at this time is interrupted after the timed reminder key is pressed.

7. The reminder call ignores the following features;
   - call forwarding,
   - hunting group,
   - call pickup,
   - do-not-disturb.
8. All station users are authorised to use the timed reminder service.

5.15 Alternating between Two Calls (Toggling)

A station user can switch back and forth between 2 connections (internal or external) as often as desired; during this procedure one party is always on hold.

- Alternating with Analog telephone between a first call (incoming or outgoing) and a consultation call
- Alternating on Digital system- or IP-telephones between
  – a first call (incoming or outgoing) and a consultation call,
  – a first call and a call taken with the call pickup key,
  – a consultation call and a call taken with the call pickup key.
- A station user with a Digital system- or IP-telephone can have up to 4 connections simultaneously (1 connection in the call condition, 3 held connections).
  The station user can alternate between the call connection and the connection last put on hold (not counting connections released during alternating).
- During alternating between parties on a Digital system- or IP-telephone, a call can be picked up using the call pickup key and transferred if desired.
- On systems with transfer by pickup, neither of the two partners of the alternating station user can pickup the call currently being held in the alternating state.
- A station user initiating a consultation call can also then alternate back to a station user on hold if the station connection from the consulted station user has been parked.
- If the waiting station user goes on-hook, the alternating station user receives busy tone when he switches back to this party.
  – The connection with busy tone is cleared down manually by the alternating station user. Following this, the alternating station user is then in a normal two–way call with the remaining station user.

- Even if the alternating station user has two or more connections on hold, after one of the held station users goes on-hook, the corresponding connection is retained (with busy tone).
- When the system recognises that a station user on hold has gone on-hook (e.g. internal connection), it clears down the connection. The alternating station user receives busy tone when he switches back. The busy tone is then ended after the alternating user switches back to other station users on hold. Further alternating in this state is ignored.

- If the current call partner of the alternating station user goes on-hook, the alternating station user receives busy tone.

  The connection is cleared down manually by the alternating station user. Following this, the alternating station user is in a normal two–way call with the remaining station user.

- When the alternating station user returns to a waiting station user who has picked up a call in the meantime, he receives music or the announcement "please wait" (Analog telephone) or a display (maximum 5 seconds) (Digital system- or IP-telephone). The alternating station user can wait until the desired station user returns again, or he can change parties and continue to talk with the other station user.
- With digital telephones, alternating is performed with a special key. With analog telephones, this function can be performed with the grounding button, with the flash key or by dialling a single-digit code.
- When the alternating station user exits an alternating call by going on-hook, he receives a recall in systems which have call pickup. When he lifts the handset he is in a normal two–way call.
- In systems with call transfer, the remaining station users are connected with one another (call transfer).
- After a consultation connection is released, the alternating station user can go into consultation again from the first call. After the first call is released, the remaining connections stay in consultation status. It is not possible to initiate further consultation in this state.
- All station users can use the "alternating between parties" feature.

5.16 Station Hunting Groups

Any stations (voice and non-voice terminals) can be combined to form a hunting group. The hunting group can be accessed via a special hunting group number, as code or call number of the leading user in the hunting group.

- Each station in the hunting group can also be accessed directly under its own call number (with the exception of the first user if his call number is the hunting group number).
- A call which arrives at a busy hunting group is placed in a queue. Incoming external calls signaled via MFC-R2 receive a busy tone if all hunting group extensions are busy.
- A call queuer with a limited number of wait memories is used for the hunting group.
  – This limit can be defined separately for each hunting group with the aid of administration and maintenance.
  – The limit can be reduced to zero.
  – The limit applies to internal and external calls.
- In the case of Digital system- or IP-telephone in a hunting group, waiting calls are signalled by a visual call waiting indication.
- Answering in the order of arrival: If there is more than one call in the call queuer, the calls are answered in the order of their arrival.
- If the hunting group is to receive emergency calls, then any emergency calls (set up with the timed hot-line service) are not rejected if all trunks are busy. Connection setup is automatically repeated on expiry of a timer.
- Indications for hunting group users if one or more calls are waiting in the call queuer (Digital system- or IP-telephone).
  - If all the station terminals in the hunting group are busy, any further calls can be distributed to the backup memories of the Digite terminals in the hunting group. When these are full, further calls are placed in the call queuer of the hunting group and are extended from there to the station terminals or backup memories as soon as they become free. The seizure sequence for station terminals and backup memories is identical.
  - The call in the backup memory is signalled by a visual call waiting indication if release for camp- on is activated.
  - If a call has been extended to a backup memory, the original assignment remains valid even if another hunting group station subsequently becomes free before the user answers.
- The hunting group is entered with the aid of administration and maintenance; the following information is entered:
  - hunting group code or call number of the leading user
  - chaining address for the stations in the hunting group
  - type of hunting group: linear hunting group or cyclic hunting group

- Linear hunting group: The search for a free station always starts with the first station in the hunting group and ends with the last.
- Cyclic hunting group: The search for a free station starts with the next station in the cyclic sequence. The search continues through all the stations in the hunting group until the calling party terminates the connection.

- If a call is not answered within 20 s, the ringing state is terminated and the call extended to another station which can answer the call. The call continues to be forwarded until a user answers or there is no further station in the hunting group which is capable of answering the call.
- Calls to a hunting group with the call number of the leading user can be passed on to another station terminal or hunting group with the aid of the call forwarding facility.
- In addition, users at hunting group stations which are not the leading user can activate call forwarding - all calls and call forwarding - no answer for their personal calls:
  - Personal call forwarding - all calls is ignored for hunting group calls
  - Personal call forwarding - no answer is ignored for hunting group calls.
- If a station does not have the class-of-service appropriate to the hunting group call, this does not affect extension of the hunting group call provided the hunting group number has the relevant class-of-service (toll access, tie toll access, DID access).
- The "release for camp-on" (call waiting - terminating) feature applies to personal calls and hunting group calls.
- Voice calling for a hunting group call
  - There is still a station free in the hunting group:
    Voice calling takes place at the first free station in the search sequence.
  - The hunting group is busy:
    The call is placed in the call queuer; voice calling is not possible.
- A user may be a member of more than one hunting group.
- A user may take his station temporarily out of a hunting group (do-not-disturb in hunting group).
  - The "do-not-disturb" state applies to all the hunting groups in which the user is a member.
  - Users can take their stations out of the hunting group regardless of whether they are the last station or not.

- Incoming exchange calls to "empty" hunting groups are then forwarded to the ATND/CAC, provided that attendant intercept has been assigned.
- Any remaining external calls in the call queue of an empty hunting group are forwarded to the attendant after a specific time (default: 3 minutes).
- Internal callers hear the busy tone. Any remaining internal calls in the call queue of an empty hunting group must be released by the callers themselves.

- If a user in a hunting group takes his station temporarily out of the hunting group, his station is skipped for hunting group calls. The next free station in the hunting group according to the rules for call distribution receives the call. A user who has withdrawn from a hunting group only receives self-initiated calls and personal calls (e.g. callbacks and timed reminders).
- Feature for a user whose call number is the hunting group number. Call forwarding - all calls for the leading user is effective for all hunting group calls.
- Callbacks, outgoing calls extended by the attendant, automatic recalls, voice calling, and timed reminders are personal calls which are not forwarded in the hunting group.
- Release for camp-on: Hunting group calls give rise to a visual call waiting indication at this station (Digite) if all the hunting group stations are busy. Personal calls camp on independently of the hunting group.
- Call forwarding - no answer

  Call forwarding - no answer activated for the station is ignored in the hunting group as long as the station remains in the hunting group. Transfer of hunting group ringing to the next station in the hunting group is performed, however. Personal calls are forwarded to the user’s personal call forwarding destination.

  Call forwarding - no answer can be suppressed via AMO.

Hunt group, networkwide

The network-wide function for a hunting group is based on expanding overflow destinations and applying call forwarding for this feature.

1. Overflow destinations

   Calls from a hunting group can overflow to internal, network-wide and public users, as well as to VoiceMail internally and network-wide.

   Hunting group overflow has the same consequences as call forwarding, i.e. call forwarding - no answer and call forwarding on busy can not be carried out during an overflow. Linking call forwarding is only possible if the system permits double or multiple call forwarding and if this link forwards calls to a user within the node.

The are only exceptions if a hunting group is also defined as the overflow destination. In this case, call forwarding - no answer is activated in the hunting group. If a hunting group has been defined as an overflow destination and a call comes back to the overflow destination, only internal call forwarding is carried out, provided that double or multiple call forwarding is permitted for the system and the 2nd or nth link is already active.

If the overflow destination of this second or nth hunting group is a voice mail server, the connection is set up regardless of whether it is an internal or network-wide destination.

2. Call Forwarding

   Calls for a hunting group member can be forwarded to all internal, network-wide and public network destinations.

   a) Linear hunting group

      Call forwarding is only useful with a linear hunting group if the last user in the hunting group chain forwards the calls. Otherwise, no users succeeding the set up hunting group can be reached as hunting group members.

      Only the last user of a linear hunting group can be permitted to forward calls to another hunting group. This ensures that calls are only forwarded to a hunting group in a second system once this last user in call distribution is busy.

   b) Cyclical Hunting Group

      Calls can be evenly distributed with call forwarding using cyclical hunting groups, if internal and network-wide users of a hunting group are defined. In contrast to linear hunting groups, any user can be specified for call forwarding when utilising cyclical hunting groups.

3. Restriction

   A hunting group can **only** be set up for users of the **same** node. The user of call forwarding allows the network-wide function.

4. Set up / Administration

   – The overflow destination for hunting groups can only be defined using AMO’s.

   – Call forwarding for hunting groups can be defined using AMOs or from a terminal (user or attendant console).

5.17 Class-of-Service Changeover

Each subscriber has two classes-of-service: a standard and an alternative class-of-service. The classes-of-service are assigned by means of the administration and maintenance operation.

Changeover between the two classes-of-service can be effected from the telephone set or from a centralised location (extension, AC, DPS, automatically by the system at preset times). If a telephone set is programmed for centralised class-of-service changeover (i.e. member of a COSX group) and COSX is not activated by locking the telephone ("Key-COSX"), it is always switched over, even if class-of-service changeover from the telephone set is additionally possible.

1. Class-of-service changeover at the telephone set

- Using a special procedure, the subscriber can change over the standard class-of-service of his telephone set to the alternative class-of-service (and vice versa).
- The subscriber receives special dial tone on lifting the handset for as long as the class-of-service is changed over.
- A maximum of 100 % of station users can be authorised for class-of-service changeover.
- The administration and maintenance system is used to determine for each telephone set whether the class-of-service changeover is effected by means of a class-of-service changeover code or with a key.
- A special code is used for identification when class-of-service changeover is carried out. However, the code can also be identical with the code used for manual input of the PIN by a mobile station user.
- The personal class-of-service changeover code corresponding to the call number is entered by means of the administration and maintenance system (a maximum of 12 digits).
- The class-of-service changeover code (COSXCD) is normally displayed in clear text on the LCD display of the user’s telephone. If required, the display output can also be suppressed, i.e. output is masked with asterisks (*). This must be set separately for each class-of-PIN (by changing a bit in the bit string).

2. Class-of-service changeover from the AC

- Changeover of the standard class-of-service of subscriber groups to an alternative class-of-service can be carried out by any operator (from the night AC).
- A class-of-service changeover which has been carried out by a user to be barred from being changed again from the AC.

- Interrogation of subscriber groups with activated class-of-service changeover.
- Change over/ switch back the class-of-service of a subscriber group.
  - By dialling the number of the subscriber group, the corresponding class-of-service is changed over or switched back.
  - Two or more subscriber groups can be changed over or switched back by successively dialling the corresponding line number and pressing the DUE key.
- Simultaneous changeover or switching back of all activated class-of-service changeovers.
- When a class-of-service changeover is activated/deactivated on the night AC, this state is signalled at the normal attendant console when operations are switched back to daytime mode.
- The administration and maintenance system is used to define the alternative classes-of-service for each individual station user.
- The administration and maintenance system is used to define the subscriber groups (max. 16); these groups are independent of other subscriber groups formed for other features.
- The subscriber for the changed over station receives special dial tone on lifting the handset for as long as the class-of-service is changed over.

3. Automatic class-of-service changeover

- The class-of-service of subscriber groups can be changed over automatically at a specific pre-programmable point in time.
- Two times (changeover/switching back) can be specified per weekday, i. e. A total of 2 x 7 different times.
- Each day the system switches automatically to the next combination of times that has been entered.
- The times can be specified individually for each subscriber group.
- The times for each day of the week and each subscriber group are defined by means of the administration and maintenance system.
- Automatic class-of-service upgrade for the night station under night circuit conditions.: Under night circuit conditions the night station is automatically authorised for override, transfer and direct exchange access.
- Automatic class-of-service upgrade in connection with the assigning/ reverting of non-dialled lines. For the assigning/reverting of non-dialled lines, the class-of-service of the subscriber wishing to have the connection is switched over automatically to the standard class-of-service pre-programmed in the PABX for the duration of this connection.

4. Transport of the class-of-service (mobile user). On a telephone set other than his own, the mobile user’s standard class-of-service goes into effect after his personal identification number has been dialled.
5. Priorities for class-of-service changeover: The class-of-service changeover or resetting which was last carried out is always the one in effect for a telephone set, regardless of where it was carried out from (from the telephone set, from the attendant console, automatically).
6. Permissible combinations of class-of-service changeovers
   - analog telephones: changeover code,
   - Digital system- or IP-telephone changeover with key ,
   - if central and decentral class-of-service changeovers are combined, the class-of-service changeover last activated is always in effect.

- All keys remain in operation following changeover to the alternative class-of-service. However, they can only be used to the degree that the alternative class-of-service allows.
- The class-of-service changeover for non-voice terminal equipment is controlled by the associated voice terminal.

5.18 Autoset Relocate

The autoset relocate feature allows some terminal types to be replugged (relocated) in the system as required, with automatic re-configuration. The feature allows users to replug their telephones as required, without having to call a service engineer for re-jumpering at the main distribution frame and re-configuring the telephone via AMO. The autoset relocate function is initiated and completed by means of a user-guided dial-up procedure with acknowledgement tones. Users with the appropriate telephones will also receive display messages.

The relocate function is only available for users with the appropriate authorisation, and must be used together with a PIN.

The relocate function can be initiated as follows:

- by telephone users, via dial-up procedure (or menu function) for their own telephone only.
- by service engineers, via dial-up procedure for any telephone to be relocated.
- by service engineers, via AMO.
- The ’log in’ and ’log off’ procedures are functionally separate procedures, i.e. a service engineer can ’log off’ and the user can ’log in’, or vice-versa, as required.

- Any user-controlled line features activated prior to logging off (e.g. call forwarding) will continue to be active after the telephone has been logged off.
- After logging off, a user cannot be reached directly until his or her telephone has been re-plugged and logged in again.
- All the user’s telephone settings and access authorisations are automatically restored after log in.
- For analog telephones, the appropriate dialling method (pulse dialling or DTMF) is detected automatically after log in.
- There is no time limit for the relocate function, i.e. log in can be carried out any time after log off.
- It is possible to initate an emergency call from a telehone which has been logged off.

Functional Restrictions

- Automatic callback requests are not possible if a user is in the process of relocating. Programmed timed reminder calls are not executed while the user is logged off, and any reminder call times which have expired between log off and log in are deleted.
- Users cannot change telephone types with the aid of the autoset relocate function.
- Users can only log in if they have previously logged off, i.e. new terminals and users cannot be introduced to the system in this way.
- Autoset relocate can only be used with telephones which can transmit dialling signals.
- Service engineers cannot initiate autoset relocate from a different telephone or via AMO if the line of the user to be relocated is busy.
- In theory, card users cannot be ’logged off’, since they are not assigned to a physical location in the system. However, if a card user uses a card telephone over a line circuit which is in logged off state (i.e. relocated user has not yet logged back on), the card user becomes a ’fixed’ user, and the circuit is programmed with the card user’s data. In order to free the circuit and the card user again, the card user must be logged off with the log off procedure of the autoset relocate feature.
- The autoset relocate function **does not** work when relocating users to other nodes of a network.

Terminals which support Autoset Relocate

- Anate telephones
- analog fax device

- Digital system- or IP-telephone (without second telephone)
  - If two terminals are connected to the same line, a strict relocate sequence must be adhered to first, the terminator must be relocated and then the repeater can be relocated
  - If a functional terminal or CorNet terminal is connected to the Digital system- or IP-telephone via an S_0 adapter, the optiset E must be relocated last.

Terminals which do not support Autoset Relocate

- HiPath Tradeboard
- HiPath Executive
- Terminals associated with other terminals
- Terminals configured under the main number of an S_0 bus, if other terminals are connected under the same number.
- Functional terminals in an S_0 bus configuration.
- IP terminals and IP softclients

5.19 Direct Station Selection

Users with an Digital system- or IP-telephone have up to 35 direct station (70 stations with restrictions) selection keys at their disposal as an alternative to repertory keys.
The HiPath 4000 system can be administered in such a way that the maximum possible 35 direct station selection keys can be entered either directly and flexibly at the terminal or, in the interests of data protection and confidentiality, **only** via AMOs..

- An internal destination can be stored for every direct station selection (DSS) key:
- The call number of a station with console answering priorities can be entered as the DSS destination (e.g. the first station in a linear hunting group, night service). Console answering priorities are ignored for a direct station selection.
- A maximum of 10 Digital system- or IP-telephoneusers can enter the same DSS destination. This can be set using administration and maintenance operations.
- A maximum of all station users with a Digital system- or IP-telephone can have DSS keys.

- The busy state of the DSS destination is continuously signalled at all Digites which have entered this destination for a DSS key by illumination of the DSS key LED.
- Automatic camp-on if the DSS destination is busy:
  - visual camp-on for Digital system- or IP-telephone,
  - audible camp-on for Anaalo telephones,
- Direct station selection disregards services activated at the DSS destination:
  - call forwarding: in an integrated executive/secretary system a direct station selection to the executive is not forwarded to the secretary.
  - station guarding,
  - call forwarding - no answer,
  - hunting group.
- Consultation can be initiated automatically by pressing the DSS key during the call.
- The DSS keys can be used to enter call numbers, e.g. for variable call forwarding.
- Transmission of a special call signal in connection with DSS can be implemented by means of the administration and maintenance system.
- Interrogation of a DSS destination
  The stored call number and the name (if available) are shown after the DSS key is pressed.
- Entering a DSS destination

  Once the call number has been entered in its entirety, the display is supplemented with the name, if available.
- Deleting a DSS destination

  The station user can delete destinations which he has entered.
- Call Pickup via DSS Keys

  Station users in call pickup groups may also use their direct station select keys (DSS keys) for call pickup within the same group (also possible within CHESE systems).

  The direct call
  - follows a RWS, respectively AUL
  - observes the Chese-function and will be signaled on the secretary-device when head absent

– can be realized in teams with up to 70 attendants, but not all direct call targets are to be inquired on one terminal

– target (name key) can be set on **any** of the keymodul

– target (DSS key) can be **only** be set on level 1, i.e. when a DSS key is used there are no additional levels available

5.20 Connection DSS1 Terminal

The S_0 bus is a user basic access with two 64-kbit/s user channels and one 16-kbit/s signalling channel (B+B+D).

In HiPath 4000 systems one of the B-channels can be blocked via AMO for single-channel operation (fixed S0 connection). .

5.20.1 S_0 Bus Functions

The public Euro-ISDN offers two ISDN connection types, which vary in detail:

1. the DSS1 passive bus connection for the connection of Euro-ISDN terminals.
2. the DSS1 System connection for the connection of Euro-ISDN PABX (see chapter 7.1 for details). In the public ISDN, the DSS1 passive bus connection is physically a basic access with S0 bus configuration.

Services and supplementary services

The following sections describe the services and supplementary services standardized for Euro-ISDN and supported on the DSS1 passive bus connection.

Bearer Services and Teleservices

On the DSS1 passive bus connection,HiPath 4000 supports the connection setup for all Circuit-mode Bearer Services and Circuit-mode Teleservices used in public Euro-ISDN.

The following Circuit-mode Bearer Services are supported:

- 64 kbps Unrestricted Bearer Service
  64 kbit/s Bearer Service
- 64 kbps Service usuable for 3.1 kHz Audio Information Transfer
  3,1 kHz Bearer Service
- 64 kbps Service usuable for Speech Information Transfer
  Speech Transfer Service

HiPath transfers all other Circuit-mode Bearer Services in transparent mode.

The following Circuit-mode Teleservices are supported:

- Telephony 3.1 kHz
- Telephony 7 kHz
- Telefax Group 4

HiPath 4000 transfers all other Circuit-mode Teleservices in transparent mode. Fallback procedures for the teleservices Telephony 7 kHz and Videotelephony are not supported.

Supplementary Services

The following DSS1Supplementary Services are supported from HiPath 4000:

MSN	Multiple Suscriber Number
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
COLP	Connected Line Identification Presentation
COLR	Connected Line Identification Restriction
SUB	Sub-Addressing
CFU	Call Forwarding Unconditional
CFB	Call Forwarding Busy
CFNR	Call Forwarding No Reply
CD	Call Deflection
ECT	Explicit Call Transfer
CW	Call Waiting
HOLD	Call Hold
CCBS	Completion of Calls to Busy Subscriber
3PTY	Three Party Service
UUSi	User-to-User Signalling Service 1 implicit User-to-User Signaling According to ETSI
AOC-D	Advice of Charge during the Call
AOC-E	Advice of Charge at the End of the Call
MCID	Melicious Call ID
TP	Terminal Portability
UUS3	User-to-User Signalling 3

The following sections contain a brief description of Supplementary Services available for HiPath 4000 DSS1 passive bus connection. For a detailed description of the implemented Supplementary Services, with options and exceptions, see the PICS forms (PICS = Protocol Implementation Conformance Statements) defined by ETSI and filled in for the HiPath 4000.

Restrictions

Certain HiPath 4000 features are not supported by DSS1 terminals on passive bus connections. The following list is not complete. In case of questions please contact HS-Door.

- ACL (application connectivity link) = Siemens specific CSTA interface):
  DSS1 stations cannot be controlled and monitored using ACL. They can therefore not be used in ACL controlled connections (either as calling or as called party).
- Hunting group:
  DSS1 stations cannot be members in a hunting group.
- Automatic recall:
  DSS1 stations release both connections - the consultation call and the call on hold - when going on-hook during consultation. There is no automatic recall.
- Transfer by going on-hook:
  DSS1 stations release both connections - the consultation call and the call on hold - when going on-hook during consultation (initiated by HOLD). The connection is not transferred.
- Night station:
  DSS1 stations cannot be used as night stations.
- CHESE(boss-secretary function), Team functions:
  DSS1 stations cannot be CHESE or team members.
- PIN-related features:
  DSS1 stations cannot use PIN-relatd features such as COS changeover, Follow me and FWD (call forward).
- ACD (automatic call distribution):
  DSS1 stations cannot be members in an ACD.
- Message waiting idication
- integrated call logging list
- Keyset 300 E functionality

Security Requirements

The MCID and UUS features are only permitted to be utilised by authorised users. The administration of user authorisations is subject to the security mechanisms foreseen and already implemented in HiPath for such authorisations.

Certifications

To maintain the reliability and high quality in HiPath networks, external Communication Servers must be certified before they can be connected to the HiPath 4000 QSIG/PSS1 interface. The certifications have to be done by a Siemens certification test laboratory. Normally, protocol tests in the laboratory and field trials are required.

Already certified foreign products must be re-certified in the following cases:

- upgrading of a HiPath 4000 to a new software version.
- if a new version of the foreing product is used.

Country-Specific Aspects

In the public ISDN of the individual countries country-specific developments of the standardized ETSI DSS1 protocolls are in use. With HiPath 4000 the DSS1 MgAs is implemented according to ETSI standards (see figures), there are no country-specific versions.

Adjustments can not be implemented.

MSN

Multiple Subscriber Number (MSN)

By means of the Multiple Subscriber Number (MSN) several station numbers can be assigned to one single port. This allows the direct calling of a certain terminal on the called port. One terminal may use several MSNs.

CLIP

Calling Line Identification Presentation (CLIP)

CLIP allows signalling and display of the calling party‘s number at the called party.

The CLIP options "no screening" ("special arrangement") on the calling party‘s side and "two calling party number delivery" on the called party‘s side are not supported by the HiPath 4000.

CLIR

Calling Line Identification Restriction (CLIR)

CLIR allows a calling party to suppress the display of his number and, if used, a subaddress (SUB) at the called party.

Suppression follows the ETSI standard and can be activated either in permanent mode or in temporary mode, i.e. on request of the calling party. The HiPath 4000 only supports the temporary mode.

HiPath 4000 supports the following default procedures for the temporary mode:

– Default value is "Presentation restricted". A user must explicitly signal, if the station number is not to be suppressed for a certain call.

– Default value is "Presentation not restricted". A user must explicitly signal, if the station number is to be suppressed for a certain call.

HiPath 4000 supports the CLIR option "Override Category". Although a calling station has CLIR, the station number and, if used, the subaddress are displayed at the called station, if the called party has the feature "Override Category".

COLP

Connected Line Identification Presentation (COLP)

COLP allows the calling station to have the number of the called party displayed.

The COLP options "no screening" ("special arrangement") on the called party‘s side and "two number delivery" on the calling party‘s side are not supported by HiPath 4000.

COLR

Connected Line Identification Restriction (COLR)

COLR allows a called station to have the display of his station number and, if used, his subaddress (SUB) suppressed at the calling party.

Suppression follows the ETSI standard and can be activated either in permanent mode or in temporary mode, i.e. on request of the called party. The HiPath 4000 supports both modes.

HiPath 4000 supports the following default procedures for the temporary mode:

– Default value is "Presentation restricted". A user must explicitly signal, if the station number is not to be suppressed for a certain call.

– Default value is "Presentation not restricted". A user must explicitly signal, if the station number is to be suppressed for a certain call.

HiPath 4000 does not support the COLR option "Override Category".
(If this option is activated, the station number and, if used, the subaddress are displayed at the calling party,although the called station has COLR.)

SUB

Sub-Addressing (SUB)

SUB allows both the calling and the called station to signal a subaddress in addition to the station number. The subaddress is then used for the selection of certain stations or for process control in DTEs.

The maximum length of the subaddress is 20 octetts (maximum length of the information element is 23 octetts).

CFU

Call Forwarding Unconditional (CFU)

CFU is used for call forwarding of all incoming calls to a station number defined by the user.

Different call forwarding destinations can be defined per MSN and per ISDN service.

HiPath 4000 does not support the CFU option, which notifies the forwarding party of call forwarding execution.

CFB

Call Forwarding Busy **(CFB)**

CFB is used for forwarding of incoming calls to a station number defined by the user, if the originally called station number is busy.

Different call forwarding destinations can be defined per MSN and per ISDN service.

HiPath 4000 does not support the CFB option, which notifies the forwarding party of forwarding execution.

CFNR

Call Forwarding No Reply (CFNR)

CFNR is used for forwarding incoming calls to a station number defined by the user, if the call is not answered within a configured period of time.

Different call forwarding destinations can be defined per MSN and per ISDN service.

HiPath 4000 does not support the CFNR option, which notifies the forwarding party of call forwarding execution.

CD

Call Deflection (CD)

A station can forward incoming calls to a different station during ringing by activating CD.

ECT

Explicit Call Transfer (ECT)

A user with a call on hold and an active call, can connect the two parties by call transfer.

CW

Call Waiting (CW)

If CW is activated a station is informed about a waiting call (no free B channel). The station can accept the waiting call, reject it (set the call to busy) or ignore it.

The HiPath 4000supports the CW option, which notifies the calling party of the waiting call.

CW can only be used for the Bearer Services Speech and Audio 3,1 kHz as well as the teleservice Telephony 3,1 kHz.

HOLD

Call Hold (HOLD)

HOLD is used to interrupt an active call and to resume it later. During HOLD a station can establish a second call or accept an incoming call.

HOLD allows the features consultation hold, brokers call and three-party conference.

With HiPath 4000 HOLD can be used for all supported Bearer Services and Teleservices.

CCBS

Completion of Calls to Busy Subscriber (CCBS)

CCBS allows a calling party to activate automatic callback if the called station is busy. As soon as the called party goes on-hook, the station initiating callback is called. When this station answers a new call to the original destination is started.

The following CCBS options are implemented for HiPath 4000:

– "Retention Timer" = 15 sec
  (time for the calling party to request CCBS, time can be configured)
– "CCBS Duration Timer" = 45 min
  (time spent waiting for the destination to become idle)
– "CCBS Recall Timer" = 20 sec
  (time for the calling party to accept the callback, time can be configured)
– "Destination B idle guard Timer" = 10 sec
  (time until the called party is signalled idle)

3PTY

Three Party Service (3PTY) - three party conference

A station with an active call and a call on hold can interconnect the calls to form a three-pary conference. The active call and the call on hold can be incoming or outgoing. The convening party can release, or go into private communication with one of the parties. Each of the other parties can end his/her conference connection.

HiPath 4000 does not support the option intended in the standards which would allow the two other parties to put their connection to the three-party conference into hold.

3 PTY can only be used for the Bearer Services Speech and Audio 3,1 kHz as well as the teleservice Telephony 3,1 kHz.

UUSi

User to User Signalling Service 1 implicit (UUS1i)

UUS1i allows the D channel to be used for transparent bidirectional transmission of a limited quantity of user data between two stations during call setup and call cleardown.

The capacity of the data transmitted in the ’User-User Information Element’ of the SETUP message is extended from 128 up to 131 octets in HiPath 4000 (as defined in 1TR67).

AOC-D

Advice Of Charge during the Call (AOC-D)

If AOC-D is activated, the accumulated call charges are transmitted to the station either as call charge units or amounts with currency designation during the active call and at the end of the call.

HiPath 4000 provides permanent call charge information on DSS1 terminals with AOC-D classmark. Activation/deactivation of AOC-D from the station is not supported.

AOC-E

Advice Of Charge at the End of the Call (AOC-E)

If AOC-E is activated the total amount of accumulated call charges (call charge units or amounts with currency designation) is displayed at the end of the call with the first release signal.

HiPath 4000 provides permanent call charge information on DSS1 terminals with AOC-E classmark. Activation/deactivation of AOC-D from the station is not supported.

MCID

Malicious Call ID (MCID)

The MCID feature enables a called party to request the identification of the calling party.The incoming call is identified in the exchange. There the calling party number, the date and the time are recorded and printed out by connection.

The called party can initiate "identification" as follows:

– Prior to accepting the call during ringing
– During the call
– Shortly after the calling party goes off-hook , before the called party has gone off-hook.

"Shortly" means:

a period of time that can be adjusted, default as in the respective public ISDN

Identification in HiPath 4000 is only possible if the phone number of the calling party is signalled by the superordinate network. This is not the case for calls from the public telephone network if the caller uses CLIR and 'override category' is not set up in the public network for the HiPath system. In this case HiPath 4000 identifies at least the trunk group or the routing code.

Identification cannot be prevented by the calling party's CLIR supplementary service.

It is to be possible to identify the malicious caller either (system parameter in HiPath 4000):

– in the public ISDN for incoming calls that originate in the public network
– in the HiPath system of the called party for incoming calls that originate from the private network or from external networks including the public telephone network.

TP

Terminal Portability (TP)

The Terminal Portability LM makes it possible for a user to move his terminal unit during a call from one terminal box to another on the same S_0 passive bus connection. This feature also enables a user to transfer a call from one terminal unit to another on the same S_0 passive bus connection .

The portability of a terminal unit in the idle state is a constituent part of the basic access properties and does not require a procedure.

It is not possible to move a terminal unit during the calling phase and during the clearing phase.

The activation of this service is acknowledged to the user who activated it, and signalled to the other party. If the other party is an analogue terminal unit, the signalling information is ignored in the other party’s exchange.

TP are not supported on Terminals connected via adapter PNTQ

UUS3

User-to-User Signalling 3 (UUS3)

The UUS3 feature enables a user to exchange a certain amount of information with the other party during an active call. During this process, UUS3 is explicitly requested by means of indicators, confirmation by the remote party is also required.

UUS3 is a guaranteed bearer service, i.e. if the feature is not supported, the user making the transmission is informed.

UUS3 can be requested by the calling party during call setup or during an active call. The called party can only request UUS3 during an active call.

UUS3 can be requested as "essential" or "non essential". "Essential" signifies that the basic call must be rejected if the requested feature is not supported. In the case of a request during an active call, UUS3 can only be selected as "non essential".

In Germany the public network limits the number of User Information messages per direction to 16 messages per 10 sec.

UUS3 is used in conjunction with all circuit-switched services that are supported on DSS1 passive bus connection by HiPath 4000.

5.20.2 Telefax Features

Group 2/3 facsimile devices can use a telephone to control the connection setup. This telephone can be either an analog telephone or a digital system telephone. There are facsimile transmissions which interrupt voice connections and speech-accompanying simultaneous facsimile connections.

- A fax connection can be set up selectively via an associated telephone or established by switching over from an existing call.
- In the speech connection which precedes a fax connection, all features for voice service are in effect until the changeover takes place.
- If a fax connection is set up from a digital system or analog telephone without a speech connection being set up beforehand, the destination call number must be entered following the fax code. If the station user wishes to establish the fax connection from the call state, the fax code only without call number is entered and the fax device started.
- Associated dialling on a digital system telephone:

  - A fax connection from a fax device to a adapter is carried out using the non-voice (NV) key and is possible in all call switching states (simultaneous fax connection).
  - An incoming fax connection to a fax device on a adapter is indicated to telephone by the flashing non-voice key; the fax device is called at the same time . No fax connection is possible if the NV key is seized. When the flashing NV key is pressed, the telephone display indicates that the fax device should be switched to receive.
- On fax devices with automatic call acceptance, call signalling on the digital telephone (NV key) is ended when the fax device answers.
- Associated dialling on an analog telephone:
  - Fax connection setup by Analog telephones is effected by dialling a special fax number using a fax code so that a fax connection can be distinguished from a possible voice connection at the same terminal.
- Self-dialling fax devices are handled as devices with associated dialling by analog telephone.
- The following features are available for fax connections:
  - classes-of-service (fax service-specific)
  - direct inward dialling (incoming external traffic)
  - setup of outgoing external calls (direct dialling and inter-Communication Server traffic)
  - internal calls
  - speed calling - central
  - speed calling - individual (not for DSS1/Q.931 devices on S_0 bus)
  - repertory keys (for dialling on associated telephone)
  - hunting group
  - multiple trunk group
  - call forwarding (based on directory number and equipment)
  - transfer of ringing (not for DSS1/Q.931 devices on S_0 bus)
  - class-of-service changeover (not activated by terminals with Q.931 on S_0-bus)
  - user-specific numbering
  - two or more call numbers on one fax device (departmental device)
  - intercept of DID exchange calls

- hot-line service
- prevention of unallowed or undesired connections
- charge registration
- number redial for the station users (for dialling on associated telephone)
- prevention of camp-on and busy override
- callback on busy (for dialling on associated telephone)
- callback on no answer (for dialling on associated telephone)
- mailbox
- trunk queuing (for dialling on associated telephone)
- discriminating ringing
- manual PIN (not for DSS1/Q.931 devices on S_0 bus)

5.21 PIN Network-wide / Follow-me

Within HiPath 4000, PIN functions can be carried out throughout the network. All PIN’s which can be set up in a node can be used from any node in the network. A network can have virtually any number of nodes and thus any number of PIN’s (however, the node number is limited to 3 digits).

Functions / Restrictions when using a PIN network-wide / external

Feature	Network-wide / external PIN
Transfer authorizations	For outgoing calls only, not for incoming calls. Authorizations for trunk and tie traffic only.
PIN for call detail recording	Yes
Follow me / FWD transfer. Caution: Note the restrictions for FWD and FNA	Yes
Speed dialing facility	The speed dialing facility is not transferred. The speed dialing facility index from the physical station and numbers from the local system list are used.
Individual speed dialing	No. Individual speed dialing cannot be used.
WABE	Only numbers from the WABE group are transferred. The corresponding WABE group of the external system is then valid. In other words, WABE groups or the generation of systems in the network should be consistent.
IBM SMART, CF switch	Yes (for virtual and physical stations).

Feature	Network-wide / external PIN
ITR group	Only numbers from the ITR group are transferred. The corresponding ITR group of the external system is then valid.
Repertory dialing keys	Repertory dialing keys are not transferred. The repertory dialing keys on the physical station are used.
DSS keys	DSS keys are not transferred. The DSS keys on the physical station are used.
Redial	Not transferred. The redial key on the physical station is used.
Enter timed reminder	No. The timed reminder is valid for the physical station.
Mailbox	Signaling via LED only, no scroll function available. In other words, you must use the corresponding VMS code to dial into the VMS. Although the physical station's callbacks are signaled, it is not possible to scroll through or perform callbacks when the PIN is active.

5.21.1 Manual User Identification with PIN

5.21.1.1 Personal orientation

HiPath 4000 is a personally oriented switching system. Classes-of-service, charging, internal traffic restrictions, speed calling destinations, name keys, mailboxes, timed reminder service, and many other features can be personally oriented.

The feature manual PIN user / mobile user is used for personal identification purposes.

1. For a station owner at his own terminal.
   This gives the station owner access to features barred to all other users due to changeover to a lower class-of-service.

2. For mobile users.
   If an outside user identifies himself at a station, he can use the same classes-of-service and options that are available to him at his home station on the basis of his standard class-of-service.
   The "standard" class-of-service is the COS1 class-of-service assigned to the user by the system administration.

When the manual PIN user /mobile user feature is activated, the personal feature authorisations and group assignments such as dial plan group, ITR group, speed dialling list etc. of the user with the manual PIN or chip-card are used for connection setup and feature calls. In particular, call charges are also assigned to the account ("home" station) of the identified person.

The options offered by HiPath 4000 to station owners and to mobile users who identify themselves with ID cards apply equally to users who identify themselves with manual PIN.

5.21.1.2 Terminals used to activate PIN

The manual PIN procedure can be entered at Analog telephones and Digital system- or IP-telephones

Personal identification with manual PIN is valid for the device at which the PIN is entered and for the associated devices of the station used.

These terminals can be used to select activation and deactivation of the identification procedure.

5.21.1.3 Input priorities

If a manual PIN entry is made at a digital system- or IP-telephone when an ID card has already been inserted, the manual PIN entry is rejected (except for short PINs).

The PIN entry at a non-voice terminal is not possible, but only via the associated telephone.

If a new PIN entry is made after a previous PIN entry at the same station or at the associated terminals, the old PIN is overwritten by the new one.

If an ID card is inserted after manual PIN has been activated at the telephone, the ID card is valid and manual PIN is overwritten.

For all activities at a terminal the PIN valid at the time of the activity applies for the terminal used.

An exception to this is call charging in which the PIN that was activated at the time of call setup remains in force for the entire duration of the call section, even if the PIN is erased or overwritten by a new PIN during the call.

Erasing the PIN during a call does not clear down the call. i.e. any activated feature, such as call forwarding, remains activated even if the corresponding feature COS has been cancelled after the PIN timer has expired.

5.21.1.4 Types of PIN

The types of PIN are AMO-defined via a class-of-PIN parameter (COPIN), and are described in detail in the following chapters.

PIN numbers are either shown in clear on the terminal display when entered, or as asterisks (masked entry).

The PCODE number is not assigned to a specific user. A valid PCODE entry is acknowledged by the display "IDENTIFICATION + <number of the station used>".

The standard COS (COS1) of the physical terminal is used.

Manual PIN user / mobile user

1. Classmarks of user's COS1 apply.
   - as long as a PIN is active, the classmarks of the user’s standard class-of-service as defined per AMO in the user’s COS1 apply.
   - Certain key functions of the user's "home" station can be downloaded, especially repertory key functions and LEDs, as far as possible without changing the functional layout of the foreign station:
     – repertory keys and DSS keys, provided these are in the same place
     – Mailbox key / LED function
     – call forwarding key
     – saved number redial key
     – timed reminder key (timed reminders can be programmed and erased from the foreign station)
2. The PIN remains active for a predefined period of time. Default: with timer.
   - Activate: via procedure, with PIN entry
   - Deactivate:
     – via procedure, without PIN entry
     – via administered timer.
   - Existing manual PIN activations / mobile user activations are overwritten by each new activation procedure for:
     – another manual PIN / mobile user activation,
     – a long PIN for company calls
     – an ID card PIN.

     The original PIN is deactivated.
   - The following PINs can be activated without overwriting an existing manual PIN / mobile user PIN:
     – short PIN for company calls and a
     – short PIN for private calls

     When the "short PIN" is deactivated, the telephone returns to the same PIN status as before (i.e. no PIN activated or long PIN activated).

In case a user forgets to deactivate a long PIN via the manual PIN procedure, there is a time supervision system. Manual PIN is deactivated automatically after a certain time which can be set with the administration and maintenance system. The supervision time is discontinued during seizure and restarted at the end of the seizure.

Long PIN - company calls

1. Classmarks of user's COS1 apply (standard COS).
2. The calls (e.g. company calls) are marked with a code in the call data recording system.
3. The PIN remains active until deactivated at the foreign station. Default: no timer.
   - Activate:
     – via procedure, with PIN entry
     – by inserting the PIN card.
   - Deactivate:
     – via procedure, with PIN entry
     – by removing the PIN card.
   - Existing manual PIN activations / mobile user activations are overwritten by each new activation procedure for:
     – another manual PIN / mobile user activation,
     – a long PIN for company calls
     – an ID card PIN.

     The original PIN is deactivated.
   - The following PINs can be activated without overwriting an active manual PIN / mobile user PIN:
     – short PIN for company calls and a
     – short PIN for private calls

     When the "short PIN" is deactivated, the telephone returns to the same PIN status as before (i.e. no PIN activated or long PIN activated).

Short PIN - company calls

- Classmarks of user's COS1 apply for the duration of one single call, following which the foreign station is reset to its original state (i.e. no PIN activated or long PIN activated).
- The calls (e.g. company calls) are marked with a code in the call data recording system.

- The PIN remains active for one call only. Default: active for one call only.
  - Activate: via procedure, with PIN entry
  - Deactivate: automatic deactivation when call is terminated.

  The PIN can also be activated while a ’long PIN’ is active, i.e. without overwriting an active ’manual PIN / mobile user PIN’ or ’long PIN - company calls’.

  Since the various PINs can only be activated when the telephone is idle, it follows that no other PINs can be entered while the short PIN is active. When the "short PIN" is deactivated, the telephone returns to the same PIN status as before (i.e. no PIN activated or long PIN activated).

Short PIN - private calls

- Classmarks of user's COS1 apply for the duration of one single call, following which the foreign station is reset to its original state (i.e. no PIN activated or long PIN activated).
- The calls (e.g. private calls) are marked with a code in the call data recording system.
- The PIN remains active for one call only. Default: active for one call only.
  - Activate: via procedure, with PIN entry
  - Deactivate: automatic deactivation when call is terminated.

  The PIN can also be activated while a ’long PIN’ is active, i.e. without overwriting an active ’manual PIN / mobile user PIN’ or ’long PIN - company calls’.

  Since the various PINs can only be activated when the telephone is idle, it follows that no other PINs can be entered while the short PIN is active. When the "short PIN" is deactivated, the telephone returns to the same PIN status as before (i.e. no PIN activated or long PIN activated).

PIN on chip-card (only digital system telephones with Card reader)

- As for "Manual PIN User / Mobile User"
- The PIN remains active until the chip-card is removed from the card reader.
  - The chip-card must be inserted when the telephone is idle; if a chip card is inserted into the card reader while the telephone is in any other state, the system will ignore the chip card.
  - If the chip-card is removed during a call, the PIN will remain active until the call is terminated.
  - An active chip-card PIN blocks all other long PINs, i.e. no other PINs can be entered while the chip-card remains in the card reader, except for short PINs.

- Deactivate: only by removing the card from the card reader.

PCODE (Project Code)

- The project code applies for the duration of one call. Default: short PIN.
  – Activate: via procedure, with PCODE entry
  – Deactivate: automatic deactivation when call is terminated.

  The default value can be changed to "long PIN with/without timer". The PCODE can then be used in the same way as these PINs (see above).
- The classmarks of the foreign station's COS1 (standard COS) are used.
- The PCODE always remains active for all call segments, e.g. if the call is transferred to a different station.
- When the call is terminated, the foreign station is reset to its original state. When the PCODE is deactivated, the telephone returns to the same PIN status as before (i.e. no PIN activated or long PIN activated).

5.21.1.5 Several PINs at one home station

Normally, each PIN in HiPath 4000 is assigned to a separate home station. It is also possible to assign several PINs to a single home station. Several users with different PINs can then identify themselves in HiPath 4000 both as owners at their home station and also as mobile users. In the call processing system these users are treated as one person; only in call data registration and in connected servers are the individual PINs evaluated.

5.21.1.6 PIN users without home station

HiPath 4000 allows more users to be assigned than the number of telephones/ terminals available. The system can thus administer permanent mobile users, with individual classes-of-service and features. These users do not have a physical home station, but only a virtual one. It is also possible to assign several different PINs to a single virtual home station.

5.21.1.7 Identical PINs

HiPath 4000 also allows identical PINs to be used simultaneously at different stations. If several users are to have the same classes-of-service and the same facilities as a single person - for example a team - the team members can identify themselves as mobile users or as owners at the home station with a single PIN (caution: individual telephone key layouts can be permanently overwritten if downloaded from a user’s home station).

5.21.1.8 PIN code length

The length of the PIN code for the PIN types "manual PIN code" and "PIN on chip-card" can be set separately for each HiPath 4000 system,up to12 digits.

The number of characters in the manual PIN may be different from the number of characters in the class-of-service changeover code and the PIN with ID card. Different PIN types may also have different PIN lengths.

5.21.1.9 PIN code verification

How the PIN is displayed on the terminal's display during PIN entry depends on a bit which can be set per Class-of-PIN in the feature bit string. If the bit string is set to "masked display", each code digit entered will appear as an asterisk.

The PIN entered by the user is verified. If too many characters are entered the PIN is rejected.

A PIN code entered at a station is searched for in a PIN code table containing all the codes permitted in that particular system. If the code is there it is then checked to find its home station.

If the home station address is the same as the station at which the PIN was entered, the owner has identified himself and the PIN code is accepted.

If the home address of the PIN code is not the same as the address of the station used, the user identifying himself will be an outside user (mobile user). In this case a check is made to find whether this user is authorised to identify himself at this station and then use it as his own.

For this check, all HiPath 4000 users are divided into groups, there being a maximum of 16 groups. A matrix created customer-specifically with the administration and maintenance system is used to verify whether the group to which the person identifying himself belongs matches the group to which the owner of the used station belongs. As the matrix is a full matrix, verification can also be directional (A can identify himself at B, but not B at A).

It is possible to deactivate checking of the PIN/project code. By doing this, registration is carried out only for the purpose of call charge registration. The length of the PIN/project code (number of digits) is permanently set in HiPath,. This utilisation does not affect the user procedures for PIN/project code entry.

- Operational initialisation of the variant PIN/project code call charge registration without PIN/project code checking is implemented according to the following procedure:
  - Initialisation of a concrete or virtual user (any user), not necessary for PCODE,

- Initialisation of the PIN/project code length requested by the customer (number of digits)
- Set classmark "no verification" in COPIN (class of PIN).

During all activities after PIN/project code entry without PIN/project code checking, the data/characteristics/ classes of services of the station used are taken into account along with their standard class of service (upper class of service). After the PIN/project code is entered, the pin/project code and the number of the station used are recorded during call charge registration without PIN/project code checking.

- Administrators can block the use of a PIN except at the user's home station by AMO.
- The system can also be set via AMO to record wrong PIN or PCODE entries. In this case, each sixth wrong attempt (default value) from the same station causes an immediate output to the line printer.
- For each PIN type, administrators can define via AMO whether or not the PIN entry acknowledgement display "IDENTIFICATION + station number" remains on the station display for the entire duration of the PIN activation, or only for a short time.
- PIN numbers are only recognised in the user's home node. For call data recording purposes, however, they may be transferred to a call charge computer in another node.

5.21.1.10 Differentiation of features

The features manual PIN, class-of-service changeover (COSX), PIN with ID card (IDCR), and project code (PCODE) are linked together and related to each other.

- Class-of-service changeover (COSX)

  Each user has two classes-of-service, which contain separately-defined classmarks for all services:

  - a standard class-of-service (COS1) and
  - an alternative class-of-service (COS2).

  The class-of-service changeover code can be used to switch between these two classes-of-service.

  This class-of-service code can be the same as a user's manual PIN code or ID card code, but can also be different if there is greater need for security with manual PIN or PIN with ID card.

A station owner can only carry out procedural class-of-service changeover for his own station. It can be accomplished without manual PIN or PIN with ID card being present. It can also be accomplished if the station owner's PIN is present, in which case the class-of-service changeover code need not be entered.

Class-of-service changeover is rejected if an outside user's PIN is active.

On identification with manual PIN or PIN with ID card, the normal class-of-service always applies irrespective of the class-of-service currently set via class-of-service changeover. After manual PIN has been deactivated or the ID card withdrawn, the class-of-service set by the station owner's class-of-service changeover feature applies once more.

Procedural class-of-service changeover is only accomplished at telephones, but is effective for all non-voice devices associated with the telephone terminal.

- Identification with ID card

  PIN with ID card has priority over manual PINs. This means that when the ID card is inserted, a previously active manual PIN is overwritten, and a manual PIN entry made while the ID card is inserted is rejected (except for ’short PINs’). Nor can the PIN be deleted manually if it was entered with the ID card.

  The number of ID card PIN characters is the same for all ID cards in each system but can be different from the number of characters in manual PIN and the class-of-service changeover code.

- Account number input with project code

  The "account number" feature is intended to allow project-specific call charges and feature authorisations to be assigned to a project code (PCODE), as opposed to a user's home station. Several different PCODEs can be entered at one station; however, due to the need to prevent misuse as far as possible, it does not follow that one PCODE can be entered at several different stations.

  The project code procedures are the same as those of manual PIN.

  Mobile project code entry can be prevented with internal traffic restrictions for mobile users, in the same way as manual PIN and PIN with ID card.

5.21.1.11 Related features

Together with other features, manual PIN and PIN with ID card provide users at their home stations with extended options by raising the class-of-service or increasing operating convenience, and enable "mobile users" who can use other stations individually in the same way as their home stations.

In most cases following PIN entry, the key functions and LED statuses of the user's home station are downloaded to the foreign station. Administrators can deactivate the automatic downloading by means of an AMO, with the exception of call forwarding destinations, which can always be downloaded.

1. Call forwarding - all calls

   Every user can activate call forwarding (FWD) - all calls for each service separately from the device for the service in question at which the user without FWD is called to a destination device of the same service or to a device which enables interworking with that service.

   This applies to all stations at which no PIN, or the station owner's PIN, or the PIN of a co-user of the station has been entered.

   For non-voice terminals, co-users of a station are those users whose call number is normally with other call numbers at a different device (departmental device).

   A user who identifies himself at a station of which he is not the owner or co-user will be a mobile user. He can activate (transport) call forwarding from his home station to the other station at which he has just identified himself with an input procedure at the other station, for each service separately.

   If any preset or variable FWD has been activated procedurally at the home station or by transport from the other station, this FWD can be deactivated from the other station separately for each service.

   If FWD is activated/deactivated at the other station (transport/end of transport), this is signaled at the home station as if the activity had taken place there.

2. Call charges

   In call data registration the max. 12-digit personal identification which is part of the call data record is stored. It is set to zero if no PIN was present at the start of the chargeable call.

   - If call detail recording takes place in a HiPath node other than that at which the charge-originating user is located, with S_0/S_2 networking with CorNet-NQ, the PIN/PKZ is transmitted to the node at which call detail recording takes place.

   - Charged calls by PIN users from foreign stations always appear in the call data record followed by the user's home station number. Administrators can set this function to record the number of the foreign station instead.

   - In the case of PCODE calls, the call charges will continue to be allocated to the project code even if the call is transferred by the caller to a called party with an active PIN.

- In the case of exchange calls set up by a PIN user from a foreign station, the PIN number remains active for that call even if the call is transferred, unless the transferred party or called party is also a PIN user at a foreign terminal, in which case the PIN of the original caller is deactivated and the new PIN becomes active.

3. Classes-of-service

   - No PIN present:
     The currently set class-of-service of the owner of the station being used applies.
   - Owner's PIN present:
     The station owner's normal class-of-service (COS1) applies, irrespective of which class-of-service is currently activated.
   - Other PIN present:
     The normal class-of-service (COS1) of the person who has identified himself applies, irrespective of which of his classes-of-service is currently activated at his home station (outgoing calls only).

   Exception:

   If a destination user's class-of-service is used for connection setup or feature control, the effect of any activated PIN is as follows:

   - No PIN or outside PIN present:
     The currently set class-of-service of the destination station owner applies.
   - Owner's PIN present:
     The normal class-of-service of the destination station owner applies, irrespective of which class-of-service is currently activated.

4. Name keys (repertory keys)

   The digital system- or IP-telephone has name keys. The assignment of function keys as name keys and their destination call numbers is programmable.

   - No PIN or owner's PIN present:
     The name keys programmed by the station owner apply.
   - Other PIN present:
     If the station has a name key in the same position as the name key actuated on the other station currently being used, the destination call number entered at the home station applies.

   This applies to connection setup and to entering, erasing, and interrogating the destination call number.

   If the home station does not have a name key in the same position as the name key actuated, the destination call number programmed by the owner of the station being used applies.

The PIN or PCODE can also be programmed on a repertory key or in an individual speed dialling list. Default value: PIN is not programmed on repertory key or in speed dialling list.

Tthe feature code can also be stored on the same repertory key as the PIN/ PCODE.

5. Direct station selection

   - No PIN or owner's PIN present:
     The direct station selection keys set up by the station owner apply.
   - Other PIN present:
     If the home station has a direct station selection key in the same position as a direct station selection key actuated on the other terminal currently being used, the destination entered at the home station applies.

   This applies to connection setup and to entering, erasing, and interrogating the direct station selection destination.

   If connection setup is carried out from the other station, this is signaled to the destination as direct station selection. The user of the other station does not however receive direct station selection signaling (busy status) as at his home station.

   If the home station does not have a direct station selection key in the same position as the direct station selection key actuated, the destination programmed by the owner of the used station applies.

6. Number redial

   Every person has a common number redial memory for voice, and non-voice.

   - No PIN or owner's PIN present:
     For storing, redialling, interrogating, and erasing the destination call number the station owner's number redial memory is used.
   - Other PIN present:
     For storing, redialling, interrogating, and erasing, the number redial memory at the home station of the person who has identified himself is used.

   A destination call number stored at the other station can be redialled, interrogated, and erased at the home station.

7. Mailbox

   The "mailbox" feature is used to indicate to a person that there is a callback request or a message in a server. The person can use a Digite to scan the mailbox, retrieve the message, have the message output at a terminal of a service appropriate for the message, or erase the message.

   - No PIN or owner's PIN present:
     New messages are stored, and existing ones scanned, retrieved or erased with the mailbox memory of the owner of the used station.

- Other PIN present:
  Scanning, retrieving, and erasing is carried out in the mailbox memory at the home station of the person who has identified himself. New messages are stored in the home mailbox memory of every person irrespective of whether and where they have identified themselves at other stations in the system.

If a home station only has an Analog telephone, scanning, retrieval, and erasure are not possible there. If new messages have arrived, "you have a message" is announced when the handset is lifted.

If a digital system- or IP-telephone owner has identified himself at another person's Analog telephone and there are messages in his home mailbox, he hears the "you have a message" announcement .

If transmission of a non-voice document is initiated with the start key during scanning in the mailbox, the transmission is made to the mailbox owner's home call number. This means that a mobile user at another station can use activation/deactivation of call forwarding to control whether a document is output at the other station he is using or at his home station.

Only one user can scan a mailbox at one time; more than one mobile user cannot access a common mailbox simultaneously.

8. Timed reminder service

   - No PIN or owner's PIN present:
     Timed reminders are entered, interrogated, and erased in the memory of the station being used. Timed reminders follow call forwarding if this has been activated.

   - Other PIN present:
     Mobile users can enter, interrogate, and erase timed reminders in their home memory from the other station.

   Since a timed reminder call follows call forwarding if this has been activated, a mobile user can control whether the timed reminder call is to be made at the other station by transporting voice call forwarding or at the home station by terminating the transport.

9. Speed calling

   - No PIN or owner's PIN present:
     Speed calling is used and individual speed calling entries are made, interrogated, and erased at the station as set up for its owner.

   - Other PIN present:
     Mobile users can use the speed calling destinations entered at their home station and can program, interrogate, and erase the individual speed calling destinations entered at their home station from another station.

10. Digit analysis groups

For each traffic situation there are up to 16 digit analysis groups for digit analysis so that users in different digit analysis groups may obtain different dialling results for the same dialled sequence of digits.

- No PIN or owner's PIN present:
  Digit analysis is based on the station owner's digit analysis group.
- Other PIN present:
  Digit analysis at the other station is based on the mobile user's digit analysis group stored at his home station.

11. Trunk access restriction via PIN

    Two different options exist for trunk (exchange) access restriction via PIN:

    - The PIN can be assigned a fixed exchange code, which is automatically dialled by the system when the PIN is entered
    - A fixed exchange code can be enabled by entering an assigned PIN; the exchange code remains blocked unless the PIN is entered first.

12. Internal traffic restriction (ITR)

    To prevent unallowed or undesired internal traffic, two users may only be connected together if the ITR groups to which they belong permit a connection. Each user is assigned to an internal traffic restriction group (the maximum number of groups is 16), and a central matrix created by the administration and maintenance system defines which ITR groups can communicate with each other. As the matrix is a full matrix this can be directional (A can communicate with B, but not B with A).

    If a B user programs or activates call forwarding - all calls to another B user (B'), the permissibility of this is likewise checked.

    When call forwarding is executed, the system checks on a personal basis whether internal traffic is allowed between the A end and the dialled destination call number B, before A is connected to B'. A central system check can also be made to test the internal traffic restriction between A and B'.

    - No PIN or owner's PIN present:
      – Direct connection/consultation hold:
        The ITR groups of the calling station owner and the destination call number owner apply.
      – Program/activate call forwarding - all calls:
        The ITR groups of the used station owner and the call forwarding destination owner apply.
      – Execute call forwarding - all calls:
        The ITR groups of the calling station owner and the destination call number owner apply. In addition, system data can be used to check

the internal traffic restriction between the ITR group of the calling station and the ITR group of the call forwarding destination actually obtained.

  - Transfer from consultation hold:
    The current internal traffic restriction between the owner of the waiting station and the owner of the station obtained in consultation hold applies, irrespective of the route on which the initial call and the consultation hold call were made.
  - Conference:
    Within a conference a check is made as to whether internal traffic is allowed between the owners of all participating stations, irrespective of the routes on which and the destination call numbers with which the initial call and the consultation hold call were made.

- Other PIN present:
  - Direct communication/consultation hold:
    The ITR group of the mobile user who has identified himself applies at the calling end, the ITR group of the destination call number owner at the called end.
  - Program/activate call forwarding - all calls:
    Not possible for mobile users at another station, only transport of call forwarding is possible.
  - Execution of call forwarding - all calls:
    The ITR groups of the mobile user at the calling end and of the destination call number owner at the called end apply. In addition, system data can be used to check the internal traffic restriction between the ITR group of the mobile user at the calling end and the ITR group of the call forwarding destination actually obtained (owner of the called station).
  - Transfer from consultation hold:
    The current internal traffic restriction between the mobile user at the waiting station and the person who has identified himself at the station obtained in consultation hold applies, irrespective of the routes on which the initial call and the consultation hold call were made. In addition, system data can be used to check the internal traffic restriction between the ITR group of the owner of the waiting station and the ITR group of the owner of the station reached in consultation hold.
  - Conference:
    Within a conference a check is made as to whether internal traffic is allowed between the persons who have identified themselves at the participating stations, irrespective of the routes on which and the destination call numbers with which the initial call and the consultation hold call were made.

5.21.1.12 Features barred to outside users

- Programming and cancelling preset call forwarding - all calls is prevented if an outside user has identified himself at the station in question. This applies to the preset call forwarding destinations in all services.
- Digite key functions cannot be reprogrammed if an outside PIN is present.
- Class-of-service changeover is prevented if an outside PIN is present.

5.21.2 Active PIN Recovery for SMART Operation

The SMART feature (Space, Moral and Remote Technology) is applied to telephone lines, which are not assigned to a specific user but are available for use by "mobile" or PIN users.

These users do not have fixed stations within the system. Instead, they are assigned virtual station numbers (telephone numbers) in the system, and report in to the system from a given telephone by entering a personal identification number (PIN) to which the virtual number is assigned. Since manual PIN entry is a prerequisite for SMART operation, and the features activated via the PIN must remain active for a given length of time, i.e. until explicitly deactivated by the user, the user data of the PIN user must be restart-safe.

The main features of the PIN types are:

- PIN per call (short PIN) and
- PIN per visit (long PIN, remains activated until explicitly deactivated by the user).

In order to ensure that the associated data is restart-safe as required for SMART operation, long PIN data is stored in an additional memory.

Semi-permanent memories are dynamic memories which keep their contents during hard and soft restarts, and can only be initialised by means of a reload. Data entries in semi-permanent memories are also transmitted to standby processors in the event of duplex system switchovers.

- Soft restart, hard restart and processor switchover procedure:
  Following a restart, the call processing system reads the PIN data from the semi-permanent memories and the processor executes all the functions which would normally be executed after a manual PIN entry (e.g. resetting timers).
- Reload procedure:
  Following a reload, all semi-permanent memories are re-initialised and their contents deleted. This is to avoid a deactivated PIN becoming active again due to a bit error.

- Procedure for database inconsistencies in duplex systems:
  If the database update function in a duplex system fails for any reason, i.e. the databases of the active and redundant CC halves are inconsistent, all semi-permanent memories in the database of the redundant unit are re-initialised and their contents deleted. Inconsistencies are detected by version counters in the semi-permanent memories which increment each time a database update transmission takes place. The subsequent database consistency verification simply compares the version counters.

5.22 Do-Not-Disturb

The do-not-disturb feature allows certain stations to be guarded temporarily from incoming calls. Outgoing calls from this station are still possible.

1. Variations of do-not-disturb
   - Activation from the station itself: Guards against all calls to this station
   - Activation from the AC
     – Guards a station group against all calls
     – Guards a station group against all calls from stations in this group (station-to-station restriction).
2. Do-not-disturb for a station, activated from the station itself
   - Activation and deactivation of do-not-disturb by means of standard procedures
   - The do-not-disturb feature is activated and deactivated with the do-not-disturb key.
3. Guarding of a station group against all calls, activated from the AC (controlled station restriction group)
   - The controlled station restriction group can be interrogated from any desired AC (from the night AC as well) and activated or deactivated.
   - The station groups are defined by means of the administration and maintenance system.
   - Interrogation of the controlled station restriction group state.
4. Overriding do-not-disturb
   - Authorised station users and the attendant can override the do-not-disturb function at all stations, i.e. the caller receives a normal ringing signal or busy signal and the desired station user is called if his telephone set is free.

- The authorisation to override do-not-disturb is assigned by administration and maintenance operation.
- Authorised station users automatically override the do-not-disturb of the dialled station.
- The attendant can override do-not-disturb by pressing the do-not-disturb key after dialling is completed (while a corresponding message is shown on the display).
- Urgent calls (time-dependent hot-line service) automatically override do-not-disturb. Do-not-disturb is also overridden if the urgent call has been forwarded to the protected station by call forwarding - all calls or call forwarding - no answer.

5. A call cannot be transferred to a station where do-not-disturb is activated.
   - Transfer by going on-hook : recall of the station user on hold if the station user is in consultation.
   - Transfer by pressing the UEG key the station user engaged in a consultation call remains in the consultation call state (pressing of the UEG key is ignored).
   - A station user authorised to override the do-not-disturb can transfer during the call state to a station user with do-not-disturb protection.
6. If activation or deactivation of do-not-disturb from the station itself or from the attendant console overlaps for a particular station, the procedure last carried out always remains in effect, regardless of where it was carried out from.
7. Priorities for combining do-not-disturb with other features
   - Do-not-disturb and call forwarding for a station: call forwarding has priority.
   - Call forwarding to a station where do-not-disturb is activated: do-not-disturb remains in effect as for a normal call.
   - Call forwarding - no answer to a station where do-not-disturb is activated: the call forwarding - no answer is ignored. The call continues to be signalled at the initially called station.
   - Call forwarding - no answer and do-not-disturb for a station: do-not-disturb has priority. Call forwarding - no answer is prevented.
   - Hunting group call for a station where do-not-disturb is activated: do-not-disturb is in effect for personal calls only; the hunting group call is signalled once the station is free.
   - Self-initiated call (callback, recall after going on-hook in consultation, outgoing call extended by the operator, timed reminder call): the call overrides the do-not-disturb function.

- Direct call to the station where do-not-disturb is activated: the call overrides the do-not-disturb function.
- A callback request is entered for station B where do-not-disturb is activated: do-not-disturb has priority. The callback is not carried out until the station user terminates a call after deactivating do-not-disturb (an incoming or outgoing call; the system reacts as for "callback on no answer" after do-not-disturb is deactivated). If station B deactivates do-not-disturb using the code procedure, going on-hook as termination of the procedure is used as the criterion for the callback.

8. In an integrated executive/secretary system, do-not-disturb activated for the secretary telephone system is in effect for calls to the secretary and for calls to the executive which have been forwarded to the secretary.
9. Depending on the system involved all station users are authorised to use do-not-disturb (no special authorisation)
10. Signalling
    - Internal callers who dial a station with activated do-not-disturb (DND) will receive the following signals:
      – Analog telephone users: do-not-disturb tone (Germany: ringing tone),
      – digital system- or IP-telephone users: do-not-disturb tone (Germany: ringing tone) + display output "DO NOT DISTURB"
    - External callers who dial a station with activated do-not-disturb (DND) will receive the following signals: Ringing tone. In systems with forwarding to attendant if do-not-disturb is active, these calls will be switched to the attendant console (attendant intercept feature). The display output of the attendant console always indicates the reason why the call has been re-directed (e.g. "Do Not Disturb" message).
    - If the do-not-disturb feature is active, this is indicated to the station with a special dialling tone

      If the do-not-disturb feature has been activated via the DND-key (as opposed to set up via a dialled code), the LED of the key is also lit. When the DND feature is activated or deactivated, this is also shown on the display (message "Do-not-disturb activated" or "Do-not-disturb deactivated").

5.23 Camp-On/Knocking

1. Station users who are authorised for this feature and who call a busy station can use the camp-on facility to indicate that a call is waiting.
    - Camp-on for a station.

- Camp-on for a personal call at an AC (P key).

2. During camp-on,
   - the calling party hears the ringing tone, if call waiting - terminating is activated at the called station.
   - the calling party hears the call waiting tone, if camp-on was initiated by suffix-dialling or pressing the camp-on function key.
   - digital system- or IP-telephone users will hear an alerting tone, and receive a display message.
   - Analog telephone users will hear the call waiting tone.
3. Camp-on with continuous check for free line

   If the desired party has released his line, the call is automatically switched through, provided the calling party has not gone on-hook in the meantime.
4. Camp-on takes place at the desired station.

   If an exchange call is transferred to a busy station without the camp-on (knocking) protection or do-not-disturb features, or if these features have not been activated by the user, the camp-on (knocking) or call waiting signal is always applied if
   - the recipient of the transfer is busy and if
   - the recipient of the transfer is talking, i.e. in a state where the call waiting signal can be applied.

   If the recipient does not answer the waiting call within a specified time (callback timer can be set between 20 and 40 seconds), then the transfer initiator is called back. If one of the above conditions is not fulfilled, the transfer initiator is called back immediately. An immediate callback is also carried out if the intended recipient has an analog station and is either participating in a consultation hold call, a three-way conference, or call toggling (flip-flop).

   If the call waiting signal is applied to the recipient's line, the recipient may still set up a conference call, transfer a call, and release a call from conference hold state. The call waiting signalling is simply interrupted for the duration of these transactions.
5. If the station user has activated call forwarding then the camp-on function applies to the FWD destination. Camp-on protection in these cases must be activated at the FWD destination.
6. Pickup of the camp-on call to station A
   - by going on-hook. Station A is then called. During the call, station C receives a ringback tone.
   - with call hold (digital system- or IP-telephone).

- It is then possible to alternate between the camp-on party and the party on hold.
- After the called station goes on-hook in alternating status, it receives a recall by the station which was on hold at this time.

- by clearing with a key actuation
  - TR key (digital system- or IP-telephone).
  - CL key (digital system- or IP-telephone).
  - The initial call is cleared down, and station A is connected with station C.
  - By pressing the TR/CL key during a consultation call, station A is returned to the party on hold, even during camp-on.
  - If there is a second call (without signalling) in the call queue, after the TR/CL key has been pressed the camp-on call has priority.

7. Pickup of the camp-on call as secondary call by the subscriber of the same call pickup group (digital system- or IP-telephone).

   If a subscriber receives a secondary call as member of a call pickup group, this call is signalled on the telephones of all members of the call pickup group and can be answered by any member like a first call:
     - call number and name are displayed alternately,
     - call pickup group key flickers fast,
     - delayed alerting tone.
8. During camp-on the calling party can enter a callback request; the camp-on function is then terminated.
9. Audible camp-on is possible with analog telephones
   - for stations engaged in a first call,
   - for stations initiating a consultation call.
10. Visual camp-on is possible with digital system- or IP-telephone
    - for stations engaged in a first call,
    - for stations initiating or answering a consultation call and for held stations.
11. Camp-on is prevented at A stations with analog telephones
    - when station A is not in call state (e.g. on hold, dialling, ringing),
    - at a station A or device with data protection,
      - data telephone

   - facsimile device
   - paging system
   - public address system
   - dictation access and control
   - recorded announcement service
   - entrance telephone
- at an A station with camp-on security,
- at a station which already has a call waiting
- at a station at which busy override is already active,
- at a station where the call partner is being overridden or has a call waiting,
- at a station which has initiated camp-on or override itself,
- at a station answering or held by a consultation call,
- at a station engaged in a conference,
- if station C is not authorised to camp on.

12. Camp-on is prevented at A stations with digital system- or IP-telephone
    - at a station which already has a call waiting,
    - at a station at which busy override is already active,
    - at a station which already has 2 calls (each call can comprise first call and consultation call),
    - at a station engaged in a conference,
    - if station C is not authorised to camp on.
13. Camp-on is prevented on a line to the Voice Mail Service.
14. Simultaneous camp-on/busy override to both call parties is possible.
15. Camp-on is also possible if station C (the user who has initiated camp-on) is in consultation status, and is not a cordless telephone user. For in the case of cordless telephones, only basic calls in the talking state can be camped on.
16. The administration and maintenance system is used to set
    - camp-on or, alternately, override authorisation,
    - busy override and camp-on security,
    - data protection (absolute camp-on protection for the line)

17. Temporary data protection for data traffic. The data protection is automatically extended to a data connection; camp-on can take place on these lines when they are used for voice calls.
18. The desired station (station A) has the following possibilities during the camp-on process
    - analog telephones (audible camp-on): go on-hook, initiate a consultation call, cancel a consultation call
    - digital system- or IP-telephone (visual camp-on): go on-hook, pick up a second call, transfer using the transfer key, initiate a consultation call, cancel a consultation call.
19. The call partner (station B with digital system- or IP-telephone) can activate the following features during the camp-on process:
    - call pickup,
    - alternate between two independent calls (not between the initial and consultation call).
20. A maximum of 100% of station users can have camp-on class-of-service.
21. A maximum of 100% of station users can have camp-on/override or data security.

5.24 Call Waiting - Terminating

Station users whose digital system- or IP-telephone has the "call waiting - terminating" feature key can activate the automatic call waiting feature for their station line for all external and internal calls in the busy state.

1. Activation of the "call waiting - terminating" feature is necessary if those calls which can result in a conversation call are to automatically camp on.
2. Only one call can camp on. Further calls receive a busy tone.
3. If "call waiting - terminating" is not activated at a station, the caller receives the busy tone instead of the ringing tone.
4. Camp-on with continuous check for free line

   If the desired station user releases his station line by clearing down the call or putting it on hold, the connection is automatically switched through, provided the caller has not gone on-hook in the meantime.
5. Pickup of the camp-on call at the called station
    - by going on-hook.
        – The station is then called.

- by putting on hold.
  - It is then possible to alternate between the caller and station user on hold.
  - After the called station goes on-hook during alternating between the two calls, this station receives a recall by the station user on hold at this time.
  - It is also possible to pick up the camp-on call in the conference state.
- by pressing the TR key.
  - The first call is cleared and the station user is connected with the caller.
  - By pressing the TR key during a consultation call the station user is returned to the caller on hold, even while it is camped on.
  - If a second call without signalling is in the call queue while camp-on is taking place, after the TR key has been pressed the camp-on call has priority when the calls are extended.

6. The call waiting - terminating key is also used for override/camp-on.
7. During automatic camp-on, the caller can enter a "callback on no answer"; the camp-on is thereby terminated.
8. Visual camp-on is possible with digital system- or IP-telephone
   - for stations engaged in a first call,
   - for stations initiating or answering a consultation call, and for held stations.
9. Camp-on is prevented
   - when the station with call waiting - terminating already has 2 calls (each call can comprise first call and consultation call),
   - at a station with call waiting - terminating which is already camped on,
   - at a station with call waiting - terminating at which busy override is already active,
   - at a station with call waiting - terminating which is engaged in a conference,
   - the negative acknowledgment for the caller: busy tone.
10. The called station user can go on-hook during the camp-on process, pick up the call, initiate consultation, cancel consultation and transfer using the transfer key.

11. The call partner with a digital system- or IP-telephone can activate the following features during the camp-on process
    - call pickup,
    - alternate between two independent calls (not between the first call and consultation call).
12. After call waiting - terminating, hunting group calls do not result in automatic camp-on until all hunting group stations are busy.
13. All users are authorised for call waiting - terminating.

5.25 Busy Override

1. Station users with busy-override authorisation and who call a busy station can break into the call in progress.
2. Busy override with continuous check for free line

   If the desired party has released his line, the call is automatically switched through, provided the calling party has not gone on-hook in the meantime.
3. During override the called party (station A), his call partner (station B), the overriding party (station C) and override tone are connected together (use of conference ports). All three parties can hear and speak to each other.
4. During busy override all three parties hear override tone.
5. The executive override (or call intrusion) feature can only be carried out on established connections in talking state. One of the following two variants can be set in the administration and maintenance system:
   - Variant a: standard (primary) connection
   - Variant b:
     - standard (primary) connection
     - consultation call
     - toggled calls (with one call in held state)
     - conference calls (attendants only, not for standard system users with the executive override feature)
       Attendants can intrude on a standard conference call, if the conference circuit in use is in the same system. Attendants can intrude after attempting to reach the conference user directly and after attempting to extend a call to the conference user.
     - calls to and from the attendant's personal telephone line
6. The override connection can be picked up at the wanted station as follows:

- User A goes on-hook (replaces the handset). Station A is then called. Station C gets ringing tone while the call is being made.
- User A presses the call pickup key, and the current call is put on hold.
  - It is then possible to alternate between the overriding call and the held call.
  - If the overridden party goes on-hook during alternating status, he receives a recall from the station that was held at the time.
- by clearing with a key actuation

Once the override has been terminated, the station user who initiated busy override can initiate other services (e.g. callback).

Users can only intrude on hunting group calls if all stations in the hunting group are busy, and the hunting group call queue is full (if configured). Users cannot choose which of the hunting group members to intrude on; the first connection in the hunting group in one of the above states is overridden.

Users can intrude on connections if a waiting call is being signalled (camp-on, knocking). Variant b must be configured in the A&M system in order to do this. The camp-on or knokking signal is then terminated.

Users cannot intrude on or override a connection which is already being intruded on.

7. Busy override is prevented
   - if station A is not in call state (e.g. on hold, dialling, ringing)
   - in connections where a station or device with data protection is involved
     - data terminal
     - paging system
     - public address system
     - dictation access and control
     - recorded announcement service
     - entrance telephone
   - in connections where a station with override security is involved,
   - in connections where busy override is already activated,
   - at a station which has initiated override itself,
   - at a station which already has 2 calls (each call can comprise first call and consultation call) (digital system- or IP-telephones),
   - at a station engaged in a conference,

- at a line to the Voice Mail Service,
- if the station is not authorised to override.

8. Busy override is also possible if the overriding party (station C) is engaged in a consultation call.
9. If the station user has activated call forwarding, then the override function applies to the FWD destination.

   override protection in these cases must be activated at the FWD destination.
10. Administration and maintenance operations are used to define:
    - busy override or, alternately, camp-on authorisation,
    - busy override and camp-on security,
    - data protection (permanent data protection for the station line).
11. Variant a): During the busy override process the wanted party (station A) can only go on-hook or pick up the call.

    Variant b): the wanted party (station A) and the current connection partner (station B) can use all the respective features available before the intrusion. However, this means that the intrusion call is terminated; the intruding party (station C) remains in the hold buffer and is rung when the wanted party (user A) goes on-hook.
12. The call partner (station B with digital system telephone) can activate certain features during the busy override process.
    - consultation hold,
    - call pickup,
    - alternate between two independent calls (not between the initial and consultation call).
13. For each station there is a special call queue for busy override by a station user or an attendant. This makes it possible to override a station user for which there is already a call waiting.
    - When the attendant ends the override in the course of servicing the call by leaving the line, the held call is switched to this station and is put in the call queue, if the call queue is free; if this is not the case, the call is automatically held on the call key.
14. A maximum of 100% of station users can have busy override class-of-service.
15. A maximum of 100% of station users can have busy override and camp-on security or data protection.
16. Remote override

Remote override is provided in some DID criteria schemes. If the call is not intercepted during direct dialling to a busy station, the busy tone is signalled. Busy override can be initiated on the line. Override tone is signalled. After the override, the PABX transmits music or an announcement. The line can now switch through to the caller or override again.

5.26 Emergency Intrusion / Call Terminating for Network Station User and Attendants

This feature allows the user to intrude on calls or override busy lines in the home node and in foreign nodes throughout a networked system (digital internetworking with CorNet-NQ signalling or analog networking with E&M signalling), and to terminate established connections. Manual or automatic emergency intrusion can be configured for the home node.

"Emergency intrusion / call terminating and destination-specific call terminating" is a feature intended for emergency calls in special networks (such as EPU companies, police service etc.).

This feature cannot be compared with the standard "executive override" (call intrusion) feature, which must always take activated override protection into account.

Users can only intrude on busy station lines and tie-lines **in a standard, two-party talking state** (i.e. **not** in a consultation call, conference or initial seizure state etc.).

Emergency call terminating can be carried out regardless of the call status of the busy line.

Emergency intrusion and/or call terminating is normally set up by Digite users and attendants by suffix-dialling from a switched connection state. However, users can also activate the feature before dialling the destination number (prefix-dialling).

Analog station users can only activate emergency call terminating by prefix-dialling.

Analog and digital system- or IP-telephones users and attendants can also activate a "destination-specific call terminating". Destination-specific call terminating is possible on tie lines.

The "emergency intrusion / call terminating" feature can be used in locally and in other HiPath 4000 nodes of a network, provided the network is linked via DIUC and digital lines with CorNet-NQ signalling, or TMEMW and analog lines with E&M signalling, and all nodes use the same emergency intrusion / call terminating criteria.

Emergency intrusion/terminating is only possible if the users and tie-lines have been appropriately configured. Lines can also be protected against emergency intrusion / call terminating by specifying the appropriate configuration parameters.

Emergency call terminating is carried out without warning, whether prefix-dialled or suffix-dialled.

5.26.1 Emergency Intrusion in Home and Foreign Nodes of a Network

- Authorised users and attendants can intrude on established calls in talking state if a called user is busy or if a line required for connection setup is busy.
- As soon as the called party has gone on-hook after hearing the intrusion tone, he or she is called by the intruding party.
- As soon as a busy line required for connection setup has been cleared down, it is used for setting up the connection required by the intruding party.
- The station user or line intruded on hears the standard intrusion tone (override tone).
- Connections established via the emergency intrusion/call terminating feature cannot be terminated again with this feature, but a repeated emergency intrusion on such connections is possible.
- Only standard two-party connections can be intruded on.
- Emergency intrusion on attendant calls is not possible.
- Only one emergency intrusion on the same called party or line can take place at the same time.
- Emergency intrusion can be initiated from a digital system- or IP-telephone or from an attendant console, but not from an Analog telephone.
- If a called party is in an S_0 bus configuration, the emergency override feature will only work if not more than two terminals are connected to the S0 bus.

5.26.2 Call Terminating in Home and Foreign Nodes of a Network

- Authorised users and attendants can terminate existing connections via a dialled procedure, either in prefix dialling or in suffix dialling, if a called party or a line required for connection setup is busy (a called party's connection can be terminated in all states, a busy line can only be disconnected if in a standard, two-party connection state).

- Connections can either be terminated immediately, or with previous intrusion.
- As soon as a called party's connection has been terminated, he or she is called by the terminating party.
- As soon as a busy line required for connection setup has been cleared down, it is used for setting up the connection required by the terminating party.
- Connections established via the emergency intrusion/call terminating feature cannot be terminated again with this feature, but a repeated emergency intrusion on such connections is possible.
- Call terminating can be activated from an analog-, digital system- or IP-telephone or attendant console in prefix-dialling (as a precaution), or in suffix dialling (not possible from Anates), if a called party or dialled line required for connection setup is busy. In the latter case, the line connection to be terminated is allocated by the system. The line connection is selected according to the LCR or alternate routing plans as configured for the route dialled by the caller.
- Connections can either be terminated immediately, or with previous intrusion.
- If a called party is in an S_0 bus configuration, the call terminating feature will only work if not more than two terminals are connected to the S0 bus.
- It is also possible for normal users without permanent emergency intrusion / call terminating authorisations to set up connections to certain destinations with "prophylactic" call terminating. In this case, it is the internal or external destination which is equipped with the feature; the destination number is a special code composed of the call terminating request code and the destination number itself.
  In all other respects, call terminating by unauthorised users corresponds to call terminating by prefix-dialling by authorised users.

5.26.3 Emergency Override Rejection

Emergency override is not possible in the following circumstances:

- Initiating user is not authorised
- Called party is protected against emergency override
- Shortage of system resources (no routes available)
- Emergency override is attempted on a saved two-party connection which has not yet completed recovery after a soft restart
- Emergency override is attempted on an attendant console line

- Emergency override is attempted on a line which is not engaged in a standard two-party connection. In normal circumstances, emergency override is not possible for other connection types, e.g. consultation calls. However, if emergency override is attempted in the node in which the standard two-party connection is established, and the consultation call is established to a user in a remote node, this does not apply.
- Emergency override is attempted on a line belonging to a hunting group configuration
- All trunks are busy or no trunks available which are not protected against emergency override
- Emergency override is attempted on a CHESE user’s line. If a CHESE user needs to be available in emergency situations, this can only be done by configuring an additional telephone with a separate call number.
- Emergency override is attempted on a connection which is already being overridden.
- Emergency override is attempted on a line engaged in a three-party connection.

5.27 Hotline Service

1. Immediate hotline service
   - After picking up the receiver, an authorised user is automatically connected to a certain destination.
   - Hotline can be configured for both voice and data.
   - One hotline index can be assigned to a line or trunk. For a line this index can be assigned for use with the Hotline feature or the Off Hook Alarm feature (not both).
   - Hotline for analog and digital system- or IP-telephones will be executed upon going off hook. For keysets hotline will be executed upon the line being selected and active.
   - Each of the destinations can be assigned to an ITR group (max. 16).
   - A maximum of 100 % of station users can be authorised for hot-line service.
   - A Hotline target may be:
     – A 1-6 digit extension number (or 7 digits for special countries)
     – A system speed number which is of the form, system speed access code followed by the system speed index.

- The Hotline feature of a connection can be switched on and off with a class of service changeover (centrally or with ID). Application example: Using a certain connection either as a dial connection (e.g. during work hours) or as a Hotline connection (e.g. after work hours).
- Hotline destinations can be installed individually when it has been ensured per AMO that as a destination number one of the 16 station groups in the "Hotline memory" has been provided with the code "speed dialling facility - individual" and a memory area number of 1-0. The user can individually enter each internal or external Hotline destination into the memory area preset with the digit 1-0 in the speed dialling facility.
- It is normally possible for all terminals to write in the "speed dialling facility - individual" (Hotline switched off). If the Hotline has been switched on, writing is generally not possible, since manual dialling has been blocked. Exception: digital system- or IP-telephones with a check key can always access the speed dialling facility. Stations thus equipped can change their Hotline destination at any time (without applying the class of service changeover).
- If the telephone set with hot-line service is a telephone with a dial unit and signalling keys or feature keys, then all features can be activated during the call state; this includes callback, override and camp-on.
- If hotline is configured or accessible on a telephone with a keypad and feature buttons, features can be initiated during the calling state including callback, override, and Call Waiting.
- A non-hotline call can be presented (ring) to a line or station marked as a hotline destination.
- Any number of stations can be routed to each hotline destination.
- The following are defined by means of the administration and maintenance system:
  - internal or external destinations (a maximum of 22 digits per destination),
  - assignment of authorised stations to the corresponding destination,
  - authorised stations.

2. Automatic timed hotline service (off-hook recall)
   - If an attempt is made to set up a connection but dialling is not begun or completed within a certain time, the connection is not broken; instead, the station user is connected automatically to a specific destination.
   - Off Hook Alarm for keyset, digital system- or IP-telephones and analog telephones will be executed after timeout in Dial State.

- Off Hook Alarm for analog telephones will be executed upon time-out of busy tone from A-B disconnect of one party
- Signalling of time-dependent hot-line service with an urgent call.
- Each of the destinations can only be assigned to one subscriber group (the destinations are independent of the configuration for immediate hot-line service).
- A maximum of 100 % of station users can be authorised for hot-line service.
- Any desired destination can be entered as the destination for hot-line service:
  - attendant console (attendant call),
  - station,
  - external station terminal.
- Off-hook recall destinations can be installed individually when it has been ensured per AMO that as a destination number one of the 16 station groups in the "off-hook recall memory" has been provided with the code "speed dialling facility - individual" and a memory area number of 1-0. The user can individually enter each internal or external off-hook recall destination into the memory area preset with the digit 1-0 in the speed dialling facility.
- If, in the case of time-dependent hot-line service to an external station, the trunk group is busy, the call is queued or a continuous check is made for an idle line; the station user waits with the handset off-hook.
- With time-dependent hot-line service, the call is put in the call queue of the extension if it is busy, as long as the destination for these calls is an extension without console answering priorities.
- Time-dependent hot-line service overrides the do-not-disturb feature at the destination station (even if the call is placed by means of call forwarding - all calls or call forwarding - no answer).
- Time-dependent hot-line service to an external destination can be to any desired station.
- The administration and maintenance system is used to define:
  - internal or external destinations (maximum 22 digits per destination),
  - assignment of authorised stations to the corresponding destination,
  - authorised stations.

3. Call forwarding - all calls, call forwarding - no answer and hunting group remain in effect at the destination.

Hotline

Hotline will support 512 indexes which can be assigned on a per line basis. The 512 indexes are shared between Hotline, Off Hook Alarm, and DITs.

Off Hook Alarm

Off hook Alarm will support 512 indexes which can be assigned on a per line basis. The 512 indexes are shared between Hotline, Off Hook Alarm, and DITs.

DIT (Dedicated Incoming Trunk)

DIT will support 512 indexes which can be assigned on a per line basis. The 512 indexes are shared between Hotline, Off Hook Alarm, and DITs

5.28 Voice Calling/Hands-Free Answer

- A station user with a digital system- or IP-telephone can speak directly to another internal Digite station user in the same node by way of the latter's loudspeaker.
- If the digital system- or IP-telephone called is equipped with handsfree talking, it is automatically switched on for voice calling. The called station can answer directly via the microphone.
- If the called has no handsfree talking equipment, the station user must lift the handset in order to answer.
- Station users can guard themselves against voice calling.
- Consultation is initiated automatically by pressing the voice calling key in the course of a call.
- Voice calling is only possible if the telephone set is idle.
- If the caller receives busy tone or ring tone when voice calling is attempted, he can use the normal services for these states as for a normal call (e.g. busy override, callback).
- If the caller makes a consultation call or picks up a call during voice calling, the voice calling connection is put on hold.
- Voice calling is ended
  - by release of the connection by the voice calling user,

  - after the called station user answers during voice calling; voice calling is then transformed into a normal connection.

- In the case of voice calling to a subscriber with handsfree answering facilities, both the calling and called parties receive voice calling tone (continuous tone) for 1 second.
- In order to alert a Voice Call user to the fact that his or her call is being broadcast via the called party's telephone loudspeaker, a "listener alerting tone" can be set to follow the voice calling tone at AMO-defined intervals. This can be useful if a called party answers a voice call by pressing the handsfree talking button, and not going off-hook. The alerting tone can be suppressed by the caller as required, and can be set to a duration of between 20 and 9999 ms, with a pause of 1 to 60 seconds. A patch allows the minimum duration of the voice calling alerting tone to be set to ≥500 ms.
- A station with handsfree answer is automatically released when the voice calling party goes on-hook. The station goes into idle state.
- The call is ended by the called party during voice calling by pressing the loudspeaker key (the loudspeaker is switched off, the station becomes free).
- Stop voice calling feature is activated by pressing the stop voice calling key and deactivated by pressing this key again.
- If voice calling is attempted to a station with the stop voice calling feature, the voice calling is ignored and a normal call is placed.
- A maximum of 100 % of digital system- or IP-telephone users can have a voice calling, handsfree answer and stop voice calling feature if they have the handsfree talking equipment (for handsfree answer) and the appropriate feature keys.
- A class-of-service is assigned for voice calling.
- The voice calling and handsfree answer/stop voice calling features can be assigned independently of one another for each digital system- or IP-telephone user.
- Stations with handsfree talking equipment always have the handsfree answer/stop voice calling features (stop voice calling key).
- Call pickup of a voice calling connection is not possible.
- Voice calling connections cannot be parked.
- Voice calling is not possible for a call that has been diverted (normal call to destination station even though voice calling key has been actuated).
- Voice calling in connection with hunting group
  - There is still a station in the hunting group idle: voice calling to the first idle station in the hunting sequence if voice calling is possible.

- Hunting group is busy: the call is placed in a call allotter; voice calling is not possible.

- Voice calling is only possible within the same node.

5.29 Executive/secretary system

5.29.1 Configuration of the Integrated Executive/ Secretary System

1. A maximum of 4 executive telephones per executive/secretary system
2. A maximum of 2 secretary telephones per executive/secretary system
3. Digital system- or IP-telephone can be used as both secretary telephone and executive telephone.
4. The assignment of executive telephones to secretary telephones is carried out by means of the administration and maintenance system.
5. These telephones are assigned special features associated with the executive/secretary system by means of the administration and maintenance system.
   - Call forwarding for executive calls to the secretary.
   - Call waiting – terminating at the secretary telephone for calls for the executives.
   - Calls for the executives at the secretary telephone are signalled simultaneously on the executive set. However, this signalling has a different display.
   - A callback is carried out as soon as the executive is free.
   - A callback request to the executive’s number can be entered during the call to the secretary.
6. The keys on the executive/secretary telephones can be assigned according to the configuration of the executive/secretary system.
   - Executive telephone:
     - key for deactivation of secretarial intercept,
     - direct station selection key, per executive in the executive/secretary system,
     - direct station selection key per secretary

  - call pickup key.
- Secretary telephone:
  - call pickup key, per executive,
  - key for deactivation of secretarial intercept, per executive,
  - direct station selection key, per executive.
  - deputy key.

5.29.2 Calling and Answering with the Integrated Executive/Secretary System

1. All calls for executives are normally signalled at the secretary telephone once an integrated executive/secretary system has been installed.
   - Calls to the executive are displayed in two lines on the secretary's telephone, as follows:
     - 1st line: Call number NAME (of caller)
     - 2nd line: FOR EXEC 1 (-x).

     The display attributes can be modified via AMO to read:
     - 1st line: Call number NAME (of caller)
     - 2nd line: Call number NAME (of executive)

       instead of the above.
   - Visual call waiting at the secretary telephone is only possible for one call (per executive); busy tone is applied to the next call.
   - Calls for the executive are signalled at the secretary telephone (and at the executive telephone by parallel signalling as well) only when the executive is free (i.e. when there is no call for the executive at the executive or secretary telephone and no outgoing call by the executive) or when the executive has pressed his call waiting – terminating key.
2. Parallel signalling at the executive telephone when a call for the executive is signalled on the secretary telephone.
3. Calls to the executive disregard call forwarding - all calls or call forwarding - no answer at the secretary's telephone (hunting group is not overridden).
4. Station guarding at the secretary's telephone applies both to calls to the secretary and to executive calls diverted to the secretary.

5. Direct station selection to the executive is not forwarded to the secretary. Station users who do not belong to the integrated executive/secretary system can also enter the executive’s number as a DSS destination.

6. Callback in connection with the executive’s call number.

   - Storage of a callback request on receipt of busy tone (secretary’s telephone or, in the case of deactivation of secretarial intercept, executive telephone busy).

     The callback is activated as soon as the executive telephone becomes free and the secretary has no other call for him.

   - Storage of a callback request on receipt of ringing tone (secretary or, in the case of deactivation of secretarial intercept, executive does not answer).

     – Executive has no mailbox: callback takes place when the executive goes on-hook again and the secretary has at most one other all for him.

     – Executive has mailbox: the callback request is placed in the executive’s mailbox.

   - Storage of a callback request during the call to the secretary.

     – If the executive telephone is busy, the secretary can invite the caller to store a callback request.

     – Callback is activated as soon as the executive telephone becomes free and the secretary has no other call for him.

   - When the callback request is being answered, any other call is routed to the telephone which is answering executive calls at this point in time.

7. Call signalling with 2 secretary telephones per executive/secretary system

   - Calls to the executive are signalled visually and audibly at the first secretary's telephone. They are simultaneously signalled on the display of the executive and the second secretary telephones.

   - After a timeout, an alerting tone is applied to the second secretary's telephone.

   - The alerting tone after time-out can be set by means of administration and maintenance for digital system- or IP-telephones:

     – single ringing tone, only when the terminal is in the idle condition

     – single ringing tone when the terminal is in any condition

     – triple ringing tone, only when the terminal is in the idle condition

     – triple ringing tone when the terminal is in any condition

  - no alerting tone

- The 2nd secretary telephone can also be assigned to specific executives and is then the first secretary telephone in such a configuration.

8. Following deactivation of secretarial intercept, calls to the executive are signalled directly at the executive telephone.

9. Call waiting – terminating in the integrated executive/secretary system

   - Automatic call waiting for executive calls which are signalled at the secretary telephone as long as there are no other calls to the executive or the executive has activated call waiting – terminating.

   - By pressing the call waiting – terminating key on the digital system- or IP-telephone, the secretary can ensure automatic call waiting under her own call number.

   - By pressing the call waiting – terminating key on his telephone, the executive can ensure automatic call waiting with his own call number during deactivation of secretarial intercept state.

10. Secretary answers a call for the executive

    A first call by lifting the handset.

    - A second call is answered during the call state by pressing the appropriate call pickup key. If two or more second calls are signalled, the secretary can answer them in any desired order.

    - A call is answered which is signalled on the executive telephone (after call forwarding). The secretary can pick up calls for the executive which are signalled on the executive telephone.

11. Executive answers an executive call

    - With standard digital system- or IP-telephone procedures during deactivation of secretarial intercept state.

    - With signalling at the secretary telephone, the executive can take the call by actuating the call pickup key, provided that the secretary has not answered the call (parallel signalling).

12. Executive answers calls for the secretary
    Personal calls for the secretary can be answered by the executive.

13. Secretary answers calls for other station users
    Within a call pickup group, which consists of the secretary and other station users (second secretary, another secretary, team members), calls to members of the call pickup group can be picked up from the stations in the group (including answering of calls for the executive).

5.29.3 Communication between Executive and Secretary Telephones

- Special call signal for calls from executive to secretary and vice versa
- Call to another station user in the executive/secretary system by means of direct station selection:
  - executive to secretary,
  - secretary to executive,
  - executive to another executive,
  - secretary to 2nd secretary or deputising secretary.
- Call transfer from secretary to executive
  - Transfer of a first call by pressing the DSS key and going on-hook.
  - Transfer of a second call by pressing the DSS key and the transfer key.
- A station user who has accessed an integrated executive/secretary system for consultation purposes (call answered by secretary after executive call number has been dialled) can transfer the held connection even if the secretary has entered a consultation call with the executive.
  - If the secretary "refers back", she is connected to the transferred station user.
  - If the secretary goes on-hook during the call to the executive or presses the transfer key, the transferred station user is connected to the executive.
- If the secretary extends an outgoing external call to an executive who is assigned to her (same executive/secretary group), the charges which have accrued up to this point are allocated to the executive.

5.29.4 Deactivation of Secretarial Intercept

- In executive/secretary configurations, call forwarding for calls to the executive is automatically active. The "deactivation of secretarial intercept" facility can be used to have these calls signalled directly at the executive telephone.
- Deactivation of secretarial intercept at the executive telephone by pressing the RU key.
- Deactivation of secretarial intercept at the secretary telephone by pressing the RU key.
- Deactivation of secretarial intercept can be switched on/off at the executive and secretary telephones.

- During deactivation of secretarial intercept state there is no parallel signalling of calls to the executive.
- Secretarial intercept can only be deactivated by the executive or secretary in the idle condition.

5.29.5 Deputy Circuit

- After the deputy circuit has been activated, all calls to the executive which would normally be routed to the secretary telephone are signalled at the deputy secretary's telephone set.
- Any desired digital system- or IP-telephone can be the destination of the deputy circuit (exception: executive telephone in an integrated executive/ secretary system).
- In every integrated executive/secretary system the secretary can select her own deputy.
- The destination station is entered and the feature simultaneously activated by pressing the deputy key.
- The deputy secretary must be an internal station user.
- Two or more secretaries can have the same deputy.
- The deputy circuit applies only to calls for the executive; calls for the secretary can only be rerouted to the deputy secretary by activating call forwarding at the secretary telephone.
- The DSS key for the secretary at the executive telephone can also be used for the deputy secretary. When the deputy circuit is activated, the deputy secretary is automatically entered as the destination for the DSS key.
- Signalling of executive calls at the deputy secretary telephone
  - There is no parallel signalling at the executive telephone or at the second secretary telephone while the deputy circuit is activated.
  - There is no alerting at the executive telephone if the deputy secretary does not answer a call to the executive within a certain period of time.
- CHESE users can also be members of a call pickup group. The following applies to pickup of a second call:
  - If the executive is not a CHESE user in a pickup group and has not deactivated the secretarial intercept function, second calls can be taken by the members of his pickup group.
  - If the secretary is not a CHESE user in a pickup group, second calls to a secretary's telephone that is busy **cannot** be taken by members of the same pickup group.

5.29.6 Add-On Witness

It is possible to add on a witness network-wide during an internal or external call without the interlocutor noticing. A witness can only be added on during a normal two-party call, in other words not during a consultation call, alternating or an add-on conference, etc.

- This feature can only be activated by digital system- or IP-telephones.
- Separate add-on witness key.
- The witness may be an analog user – either an individual user or one forming part of a hunting group – or an automatic user (tape unit connected to analog port).
- The witness can be added on network-wide.
- The add-on witness feature and witness are assigned by the administration and maintenance system.

5.29.7 Second Telephone

- Second telephones are digital system- or IP-telephones which form a pickup group together with a telephone in the executive/secretary system. They are used, for example, as conference corner telephones.
- Second telephones have their own call number; incoming and outgoing calls can be carried out.
- Second telephones have none of the special integrated executive/secretary system features, such as deactivation of secretarial intercept or a call pickup key for each executive.
- A call is transferred from the first to the second telephone and vice versa by means of the call park feature.
- A call to the first telephone is picked up from the second telephone and vice versa by means of the call pickup function.
- Signalling of a call for the executive on the executive and secretary telephones has priority over a call to the second telephone.

5.29.8 Private Line

A private line can be installed in the integrated executive/secretary system.

- Incoming
  - Second call number (secret DID number) for the executive.

  – Calls to the second call number are not rerouted to the secretary even if the executive has activated call forwarding to the secretary. There is no parallel signalling.

- Outgoing

  An exchange line can be reserved for the executive.

5.29.9 Ringing Tones

- ringing cadence:

  Discriminating ringing for calls between the executive and the secretary (and vice versa) by means of a special ringing cadence.

- Multi–tone ringing:

  Discriminating ringing for calls between the executive and the secretary (and vice versa) by means of a special ringing frequency.

5.29.10 Messenger Call

- A messenger is called by means of the ringer in the messenger room.
- In the case of a messenger shared by two or more executives, there is a separate ringer for each executive.
- The messenger is called by pressing the messenger key in either the idle or call state.
- After the messenger key is pressed, a special call is placed to the messenger. The call is automatically terminated after 5 seconds.
- The destination for the messenger call is entered on the executive telephone.
- The messenger call number is a 3-digit number.

5.30 Keyset

The keyset functionality enabling up to 28 multi-line appearances (one primary line and the remaining are secondary or phantom (virtual) lines. A line can appear on up to 40 different Keysets. The number of simultaneous incoming and outgoing connections on a Keyset is the same as the amount of configured lines.

This feature is only available for the digital system- or IP-telephones (also referred to in conjunction with this feature as "keyset").

Each key set is assigned a primary line, which is actually the extension number assigned to that station. These calls are also signalled on other keysets as secondary lines, appears on the assigned different key sets simultaneuosly. The keyset subscriber can retrieve or hold the calls signalled on secondary lines as well as calls on their own primary line. The phantom line can not be assigned a primary line, however, it can appear as a secondary line on any keyset.

Calls can be initiated using any line on any terminal. Depending on the system configuration, the line COS or the COS of the Keyset is used. There are, however, features which always use the COS assigned to the used terminal or the used line.

During the call, and after answering a call, the display of the called party normally displays <Station number and name> of the primary line of the calling station or the used line. The key also affects the display of the calling party after the B party answers.

Central office (CO) trunks cannot be assigned as line keys.

All Keysets where a line is assigned must be located in the same Communication Server. Network-wide Multi- line access is not possible.

LED's and Ringing

Each line (prime, secondary, phantom) has a corresponding LED for signalling and can be configured via administration to ring audibly and alert visually or alert visually only. Calls are signalled in the order they are received.

Four LED states are defined for status indication :

- Dark - Steady off
- Lit - Steady on
- Blinking fast used to indicate a ringing call
- Blinking slowly used to indicated a held call

5.30.1 Multiline Preference / Preselection

A key set having multiline appearance can be preprogrammed to select a line automatically when originating calls or answering incoming calls. A differentiation is made between automatic and manual line selection.

Automatic Line Preferences/ Selection

Automatic Line Preferences/Selection enable connection of the extension to one of the lines appearing on it, on an automatic basis when the user goes off-hook or starts on-hook dialing.

There are different options for the **preference of outgoing and the preference of incoming connections.**

A user can originate calls and choose the required **outgoing** line according to four different types of preferences:

- Prime Line Preference
- Last Line Preference
- Idle Line Preference
- No (Originating) Line Preference - A line key must be preselected or postselected each time the user goes off-hook

With **incoming** calls, depending on the configured options either a ringing, any arbitrary incoming, or no lines can be preferrentially selected. A user can accept an incoming/terminating call based on five types of line preferences:

- Ringing Line Preference
- Ringing Line Preference with prime line preferred
- Incoming Line Preference
- Incoming Line Preference with prime line preferred
- No (terminating) Line Preference - A line key must be preselected or postselected each time the user elects to answer the call

Signaling of incoming calls on the prime line and secondary lines:

Incoming calls are displayed when the keyset is idle and when the line is a ringing appearance. When the keyset is busy or when the service menu is in use , then incoming calls are not shown on the display. Incoming calls on non-ringing lines are displayed if the terminating preference of the keyset is "incoming line preference" (with or without prime line preferred).

When the display is available for the incoming call the first call in the ring queue is shown (on first-in first-out basis and first ringing and then alerting calls basis)

If the keyset is in preselection mode, line key depressions of different line keys (ringing/alerting) show the associated calling parties.

Manual Line Preselection

Through **manual** line selection, the user can override the automatic line preference/selection by pre-selecting a line. The user invokes this feature by pressing any of the line keys prior to going off-hook. There are three configurable options. single key, preselection and post-selection.

In **single key mode** the line is selected by pressing a line key. The Keyset is switched to hands-free mode with on-hook handset.

In **preselection mode** the line key is pressed in idle state and the required line is preselected as soon as the user goes off-hook, presses either the loudspeaker, name, DSS or number redial keys or enters dialing digits. The preselection is valid for 10 seconds (configurable via AMO). If a line with ringing has been configured, the calling party is displayed when the line key is pressed until the preselection time expires. After preselection the incoming call can be received on the preselected line by going off-hook.

Subsequent line selection. The user lifts the handset and presses a line key to subsequently chose or change the line to be used.

5.30.2 General and exclusive hold.

Hold allows a user to hold a connection and then to go on-hook or to initiate or answer a call on a different line without losing the held connetion. Only two-party calls can be held. Holding of consultation calls is not possible.

If **general hold** is set, the held connection can be picked up by all Keysets where the respective line is assigned (the LED flashes slowly on those stations) by pressing the line key.

If **exclusive hold** is set, the held connection can only be resumed by the Keyset which put the line on hold (the LED flashes slowly only on this station) by pressing the line key.

There are three ways to put a line on hold: by pressing either the "Hold" or "Exclusive hold" keys or by selection from the dialog menu. Changing to another line or pressing the line key of the current line also puts a line on hold (or releases it, depending on the system configuration). It is also possible to hold a line (or release the line, depending on the system configuration) by pressing the key of the used line.

If exclusive hold is set, the Keyset which put the line on hold receives a recall with priority, if the held line is not resumed after the timeout. If general hold is set, a recall is signalled at all Keysets where the line is assigned.

5.30.3 Multiple consultation

A user with several assigned lines can initiate a three-party conference by pressing the line key of the line he wishes to override. A three-party conference is established, if a two-party call (A-B) is active on the selected line and no private call has been initiated. Conference tone is injected at all three parties (A, B, C)if multiple consultation is activated. Only one station can enter into an existing connection.

5.30.4 Privacy calls

There are two types of privacy calls:

Automatic Privacy

A station with the Automatic Privacy feature has **exclusion** activated whenever it makes a call. This feature gives a Keyset user security against another Keyset user attempting the Bridged Call feature. Privacy is automatically activated for originating and terminating calls.

Manual Privacy (Privacy On/Off)

The Manual Privacy feature permits a station user without the station authorization’automatic privacy’ to invoke privacy by activating the feature and excluding other stations from bridging on to a line appearance of the line to which the station is connected.

5.30.5 I-Use (I am using this line) Indication

This feature provides a station user with an indication of the line to which the station is currently connected or to wich it will be connected upon going off-hook. The I-Use feature can be provided automatically as default and also manually for those stations with I-Use keys/menu item to query the line at any time.

Feature access is via an I-USE feature key, or selecting the menu item ‘’Show used line?’’ in the service menu. The automatic I-USE is active at all times upon activating, or selecting, a line.

The display is used to indicate the line active, or line to be selected, via the Automatic Line Preference function.

5.30.6 Ringer Cut-Off

A station user may disable the Keyset’s tone ringer by depressing the RINGER CUTOFF key, or selecting the menu item ‘’ Ringer Cut-Off?’’ in the service menu. All ringing line appearances, including the Prime Line, on that station will become alerting line appearances immediately.

When Ringer Cutoff is active, the LED associated with the feature key is lit, and the only indication of an alerting call is the flashing LED of the line key(s). All incoming calls will still come through but the station will not ring, only alert (the line LEDs will blink). Any call can still be answered.

5.30.7 Call origination

The System allows the user to originate a call via the:

- Off-hook
- Loudspeaker key
- on-hook dialing

- DDS key
- DSS key
- Line key
- Redial key or last number redial key
- Voice call key
- Start key (while in Mailbox scrolling mode)
- Headset Key
- Menu selection with ok-key; e.g. individual speed dial, display suppression, etc.

5.30.8 Recalls

The following recall situations are possible :

- Recall From Hold
- Recall After Transfer (Immediate, or after Time-Out)
- Recall After Camp-On
- Recall After Override
- Re-Ring

5.30.9 Audible Ringing on Rollover Lines

The Audible Ringing on Rollover Lines feature provides to hear ringing lines signaled at a user selected rollover ring volume while the user is active on a call.

Audible Ringing on Rollover Lines is used on keysets having multiple lines. It is a feature that is configured on a per keyset basis. Only lines that are assigned to audibly ring on the (idle) telephone can also be signaled when the user is active on the telephone.

This feature is associated with the multiline appearance feature.

The user configures the volume for "Audible Rollover Ring". This is accomplished via the Optiguide menus and +/- keys. For non-display telephones the +/- keys and a service code is used

Only lines configured to audibly ring on that telephone shall have the rollover ring option applied.

Provide a keyset based "signaling when station user is busy in a call " option via Administration to allow:

a) Standard Ringing at a user selected rollover ring volume on line appearances (marked for ringing) when the phone is off hook or in on-hook dial/handsfree mode.

   Note: Standard ringing equates to the ringing that would normally occur if the user were not active on a call (e.g., internal ringing, trunk ringing, etc.) or

b) Alert (Splash) Ring on line appearances (marked for ringing) when the phone is off hook or in on-hook dial/handsfree mode, or

c) No Ringing on line appearances (marked for ringing) when the phone is off hook or in on-hook dial/handsfree mode.

This feature is automatic and does not require feature access codes to activate.

One hundred percent of the keysets can be configured with a rollover ring option.

Central office (CO) trunks cannot be assigned as line keys.

All Keysets where a line is assigned must be located in the same Communication Server. Network-wide Multi- line access is not possible.

5.30.10 Preview Key for Keysets

The following functionality is part of the Preview Key feature:

- While a device is idle, it allows the user to look at the identity of the calling party on an audibly ringing keyset line before answering the line.
- When the device is active on one line, it enables the user to look at the identity of the calling party on any other audibly ringing keyset line.
- A keyset user can preview audibly ringing lines which are recalling from hold, park, system hold or transfer.
- It allows the keyset user to preview a line (configured to ring) on Manual hold
- It allows the keyset user to preview a line (configured to ring) placed on Exclusive hold by the user
- This feature will also allow the user to see the connected party information on the line the user is active on (if that line is configured to ring.
- Any attempt to preview a line that is not configured to ring, or does not meet the above criteria will result in the negative acknowledgment (timed) display "Preview of line" (line 1) "Currently not possible" (line 2), "Preview of line"(line 1) "Not possible"(line2) or "Idle".
- When the device is ringing and a line is being previewed, the currently previewed line will be selected and activated if the caller goes off-hook or presses the speaker key to go to hands-free mode.

- The preview function will be available regardless of whether the calling/called/ held/ party is a local party or a network party.
- Preview is allowed for lines configured to audible ring only.
- The feature is activated be "Preview key". There is no feature access codes available.
- The Preview display is a two-line display.

Acknowledgment Treatment

- Possible connected-party displays for line keys:
  - Calling party, held party an redirection information
  - Camp-on (call waiting) information
  - DNIS, ANI and trund-ID
  - Timed Reminders
- The previewed information display will remain visible for period set by the Pre-selection Timer. If the Preview key is pressed, when a permanent display (not timed) is on the telephone, the permanent display shall be restored after the preview display is cleared.

Time-out/Interrupt Treatment

- Display acknowledgment information due to previewing (after pressing a line key) are displayed for 1 - 30 seconds depending upon the value of the Preselection timer.

Important feature interaction and depending restrictions:

Tones, Ringing, Announcements and Displays

Push Button Dialing (non-DTMF) will deactivate preview mode.

General Features

Using the Mailbox / PhoneMail key while in preview mode will deactivate the preview mode.

Using the Volume Up / Volume Down keys while in preview mode will deactivate the preview mode and enter program/service mode.

Display Features

When the keyset is put into the preview mode, depressing of any line key preempt the existing display, if it has not already timed out.

Preview cannot be initiated during Dial State.

Common User Interface (CUI)

While actively previewing a line, use of scroll keys, or the menu while displaying the preview information will deactivate the preview display information. There are no menu or submenu items for the preview feature.

Charge Display (IM specific)

- The display information for preview overwrites the charge display. Upon time-out of the preview information, the charge display will be redisplayed.
- Using the Language Selection key while in preview mode will deactivate the preview mode.
- The Door Busy key if pressed will deactivate the preview mode.
- The display information for the Call Log feature will overwrite the Preview mode display.

Privacy Features

Using the Do Not Disturb/Ringer Cutoff key while in preview mode will deactivate the Preview mode. Using the Privacy/Privacy Off key while in preview mode will deactivate the Preview mode.

Holding Features

Using the Toggle/Connect key while in preview mode will deactivate Preview mode. When active in preview mode a user can initiate park to system. Preview mode will be deactivated.

Automatic Recall on Held Calls (Recall)

Preview can not be used to preview a recalling line that was placed on exclusive hold at a different keyset. Preview can be used to preview a line recalling from Park to Station, Park to System and from Group Park.

Call forwarding information will not be displayed upon previewing of a held line and a held line that is recalling.

Preview cannot be used to preview a line appearance that was placed on exclusive hold at a different keyset. Using the Exclusive Hold key while in preview mode will deactivate the preview mode.

Transfer Features

Using the Transfer key while in preview mode will deactivate Preview mode. Camp-on information will be overwritten by connected party information when the active line is previewed.

Call Forwarding Features

Using the CALL FWD or PROG key while in preview mode will deactivate the preview mode.

Station Pickup Features

Using the Call Pickup key while in preview mode will deactivate the preview mode

Automatic Call Distribution (Acd)

ACD provides keys for Acknowledge, Agent Message, Available, Enter ID, Primary Status, Unavailable, Work,...... In addition to their normal functioning, use of any ACD keys while in preview mode will deactivate the preview mode.

Abbreviated Dialing Features

Save/Repeat (Device)(Saved Num. Redial)

In addition to its normal functioning, use of a SNR key while in preview mode will deactivate preview mode.

While using the System/Station / DSS / DDS / LNR keys while in preview mode will deactivate the preview mode

CDR Access Codes (Account Code Input)

Using the Account Code key while in preview mode will deactivate the preview mode.

On-Hook Dialing

If preview display is active and the user is not active on a line, preview mode will be deactivated if the user initiates on-hook dialing (hot keypad dialing).

If preview display is not active and the user is not active on a line, preview mode will be deactivated if the user initiates on-hook dialing (hot keypad dialing

Using the Handsfree key while in preview mode will deactivate the preview mode when going from idle to handsfree and when going from handsfree to idle.

DSS key/ I-use

It is not possible to preview an incoming call on a DSS key. Using the I-Use key while in preview mode will deactivate the preview mode.

INTERCOM Features

Buzz can be initiated while active in preview mode. Initiation of buzz will deactivate the preview mode.

Using the Speaker Call - Two-way (Voice Calling) key while in preview mode will deactivate the preview mode.

Speaker Call - One-way BroadcastPreview is not applicable to reception of a one-way broadcast speaker call.

Service Mode

Using the Program key while in preview mode will deactivate the preview mode. Preview cannot be used to display information associated with a Night Answer call before the call has been distributed. Using the CLEAR key while in preview mode will deactivate preview mode.

Headset Operation

When active on a call Headset key will deactivate the preview mode. Auto/manual key deactivates the preview mode

Idle or Ringing: Once a line is previewed (For a Programmable display time) the user can press headset key to deactivate preview mode and answer previewed line.

Night Arrangements

A keyset configured as a night station can use the preview feature.

Difference from U.S. Implementation

There are some differences as indicated below:

– The timer for the Preview display for the Preview feature is a variable timer

– Camp-on information will not be part of the preview display information when a line key is pressed in Preview mode for the Preview.

– There will be additional negative acknowledgment displays generated for the HiPath 4000 and the negative acknowledgment display will usually be a two line display.

– There will be more feature interactions with features that exist (e.g. Timed Reminder Service, Group Park, Hot keypad dialing,...).

5.30.11 Effect on other features

The key system functionality also affects features. They are:

Dialing aids

such as speed dialing, name, DSS or number redial keys are station specific. They can be used with any line assigned to a Keyset. The LED of the DSS key shows the state of the entered destination line.

Call forwarding

Every terminal can forward all assigned lines. The LED of the forwarding key always signals the state of the primary line of the respective terminal. Forwarding of secondary lines is not signalled.

Do not disturb

If DND is activated for a line none of the Keysets where this line is assigned receive any calls. Select "Idle" for protection against incoming calls on Keysets instead of DND.

Callback

If a Keyset uses a line and enters a callback request, the primary line is displayed as callback source. As soon as callback is possible, the primary line is called exclusively. After the callback source answers, the callback destination is called non-exclusively, i.e., every Keyset where this line is assigned can answer the call.

Mailbox

Entries can be left in the mailbox of the Keyset where the used line is assigned as primary line. Browsing in the mailbox and initiation of callbacks is only possible on this Keyset,

Call waiting

is executed on seized lines. The display of the Keyset using this particular line shows the appropriate message and the call pickup LED flashes. Automatic call waiting is also executed on the Keyset using this particular line, if call waiting is allowed.

Override

is executed on seized lines. The Keyset using this particular line is overriden, the corresponding message is displayed and the call pickup LED flashes.

Paging

Every line can be routed for paging purposes and the user can answer a paging call on any line. The paging key is always assigned to the seized line.

Call pickup group

Calls within a call pickup group are only signalled (display, LED and alerting tone) on the Keyset where the particular line is assigned as primary line. Calls signalled on this Keyset can be picked up using any assigned line.

Group call

Parallel signalling is executed at specially marked stations within the call pikkup group. On Keysets group call is exclusive, which means that the call is not signalled on secondary lines and can also not be answered by other Keysets. This exclusive call can be deactivated using "Idle".

Direct call pickup

can be activated by seizing any line.

Call parking

A user can transfer the other party into the park queue of the call pickup group where the primary line is assigned, regardless which line was seized. Parked calls can be pikked up by Keysets within the call pickup group by seizing any assigned line.

Hunting group

Similar to terminals, lines can be combined to form hunting groups. Calls in the hunting group are signalled on all Keysets where this line is assigned. The hunting group key always remains assigned to the primary line of the respective terminal.

Voice calling

The called terminal is directly switched to hands-free mode, if the called party has not activated "Stop voice calling". In the case of Keysets, voice calling is executed on the terminal where the selected line is assigned as primary line. If this particular Keyset is busy, a normal call is signalled instead.

Timed reminder service

This feature is terminal-specific. It is only signalled on the terminal of the respective user, provided call forwarding has not been activated. A timed reminder service call is exclusive.

COS changeover

can be executed by any assigned line and switches between COS1 and COS2 on the used terminal.

Identification (PIN manual / ID card)

PIN activation/deactivation can be performed via any assigned line on the used terminal. The PIN is only valid for outgoing connections established from this terminal regardless of which assigned line is used. The COS (COS1) and data (e.g. name and station number) of the station for which the PIN is assigned is used. The PIN does not affect other keysets using the line of a terminal with activated PIN.

Call log

is assigned to the primary line. Only incoming calls for the primary line are entered. Outgoing calls are entered in the log of the primary line, regardless of the actual seized line.

Call charge recording

can be executed either per terminal or per line.

secretary system

Keysystem and secretary system are mutually exclusive. It is not even possible to upgrade a station intended as secretary to a Keyset using SBCSU AMO (SECR=YES).

Autoset Relocate

digital systemtelephones with Keysystem functionality can move using Autoset Relocate. digital systemtelephones (with or without Keysystem functions) can then log on at the previous port. To log on an digital systemtelephone with Keystem functions, a connection which has been prepared accordingly suffices.

5.31 Call Log for digital system- or IP-telephones

The call log records the phone-number information of the calling party in case of all incoming connections (calls or attempts). All outgoing calls' phone numbers are logged, regardless of whether the desired connections took place or not.

- A maximum of 6 outgoing and 12 incoming calls per subscriber can be stored.
- The user can browse through the log and use each entry to activate the respective outgoing connection.
- The following data are stored for each entry:
  - Phone number max. 22 digits
  - Name (if available) max. 30 characters
  - Date / time
  - Direction: incoming / outgoing
  - Info: free / busy / call
- The log is created as FIFO (first in first out memory), i.e. when the maximum possible number of entries has been reached and the next one arrives, the oldest entry is deleted.
- A call-log-key is availaible. Pressing the key the last, incoming, unsucessfull call is shown. By pressing the call-log-key or the OK-key the next entrie is shown. This function ist also available for outgoing calls.
- The call log is not possible for HiPath cordless E and HiPath Trading E.

Administration

The call log feature can be blocked or released individually for each subscriber. Two operating modes are provided:

a) logging of all incoming connections (attempts and calls) or

b) logging of unsuccessful incoming connections (attempts only).

It is possible to define the number of subscribers with a call log for each switch.

The feature can be blocked in the switch by administration and maintenance.

The call logging list cannot be altered or saved by administration

Call log is only implementet for voice services.

Impacted Features

A-User is the calling party and B-User is the called party.

Call forwarding

A-user: The destination that was dialed is entered in the log of outgoing calls, even if a different destination was reached because of call forwarding.

B-user: In the case of call forwarding (unconditional, busy), the calls are entered in the call log of the destination station. No entry is made in the log of the forwarding station.

Ring - no answer (RNA)

A-user: The destination dialed is entered in the log of outgoing calls, even if a different destination was reached because of RNA.

B-user: If after a while RNA occurs, the call is entered in the log of the originally called party (attempt) and at the time of the forwarding in the log of the RNA destination. If forwarded calls are chained, entries are made in the logs of all called parties.

Consultation holds, transfers, toggling and conference calls

In the case of multiple connections, each connection setup is treated like a separate connection. It’s adjustable which calling number should be stored in C-users call log table (A-user or B-user).

Do not disturb

If ’do not disturb’ is active at the destination station, the call is entered at the A-user and B-user as an attempt.

Unlisted subscriber / line identification restriction

A-user: The dialed destination is entered in the log of outgoing calls, even if an unlisted subscriber was called.

B-user: Incoming seizures in which the A-user is marked as unlisted are not logged.

Automatic callback

If a connection is set up as a callback, it is not entered in the call logs. However, connections are also entered in the call logs of the participating subscribers if an automatic callback is activated.

Paging

A-user: The dialed destination is entered in the log for outgoing calls.

B-user: In the case of forwarding to a paging system, the connection is not logged at the destination.

Parallel Ringing

The call logging function takes place in the log of each member

Group Call pickup / Directed Call Pick-up

A-user: The dialed destination is entered in the log of outgoing calls, even if a different destination was reached through a pickup.

B-user: If a call is picked up by a different subscriber, the call is added to the log of the originally called subscriber (attempt) **and** at the time of pickup in the log of the subscriber who picks up (call).

Parking

If a parked connection is picked up, it is added to the log of incoming connections (call) of the subscriber who picks up.

Executive/secretary

A-user: The dialed destination is entered in the log of outgoing calls, even if a different destination was reached because of distribution in a CHESE.

B-user: Calls for the executive terminal are entered in the log of the executive terminal and in the log of the secretary terminal. If a call is picked up by a subscriber other than the one dialed, the call is entered in the log when it is picked up by that subscriber (call).

Hunt group

A-user: The dialed destination is entered in the log of outgoing calls even if a different destination was reached because of distribution in a hunt group.

B-user: If a call is distributed in a hunt group, it is entered in the log of the subscriber to which the call was distributed. In the case of RNA in the hunt group, the stipulations for the conventional RNA apply.

System speed calling

If system speed calling is used, the abbreviated number is entered in the log of outgoing connections.

As the above metioned destination numbers are also kept within function keys (like DDS, DSS or redial) and there is no need to record these numbers, they are not saved in the outgoing call log. But outgoing connections through Individual Speed Dialling are logged.

Autoset Relocate

In the case of Autoset Relocate, the call log is also relocated. If a logged-off subscriber is called, the call is treated as an unsuccessful connection.

Class-of-service changeover

Access to the call log can be blocked via the COS changeover feature.

PIN / follow-me

The call log can only be read on the home device. If on a device an external identification of the PIN type ’PIN manual/follow-me’ is active or an ID card is inserted, the log cannot be read.

Voice calling

Voice-call connections are treated as normal A-User, B-User connections.

DTMF suffix dialing

Suffix-dialed digits are not logged.

Terminal failure or Blocking

If a subscriber is not available because of a temporary failure of his terminal, a blockage in the system or an overload situation, an incoming call is entered in his log of incoming connections as an attempt.

In the event of blockages in the network, i.e. if the connection request could not be signaled right through to the destination node, no entry is made at the called party.

5.32 Data Security with digital system- or IP-telephones

This feature enables information saved on digital system- or IP-telephones to be protected from unauthorised access and manipulation (and private call numbers saved on repertory dialling keys).

The following elements are provided:

- Lock for individual dialling aids,
- Lock for the service function

The locking of system-controlled dialling aids and the check function is supported on terminal units

The function (Menu key) for activating and deactivating forwarding to a stored destination, mailbox or any phone number is not restricted because these functions do not affect the terminal unit actually used.

The required locks are activated and disabled.

Dialling Aids

The access of dialling aids (Direct Destination Select / Direct Station Select and the redial key) and the usage of individual speed dialling is prevented.

Normal dialling using numeric keys is not restricted.

Service Function

As the usage of the service function is dependent on the corresponding authorisation and the appropriate user interface is to be retained, not the entire service function is locked (except for display, the check-function is locked completely there).

In the case that a authorisation is not available, only the enabled functions are provided on the menu. Direct entries into the Service menu via keys are taken into account.

The service functions can be activated, however, the scope of the functions is restricted such that the functions for displaying and changing protected data cannot be activated.

The following feature groups are affected in the user interface:

- In the Main menu, the menu item ’Reminder ’and ’Speed Dialling’ is not provided.
- In the ‘Destinations’ submenu, the following menu items are not provided:
  - ’Variable Forwarding’
  - ’Fixed Forwarding’
  - ’Speed Dial’
  - ’Redial’
  - ’Direct Destination Select’
  - ’Direct Station Select ’.
- In the ‘Key’ submenu, for the functions
  - ’Do-Not-Disturb’
  - ’Stop Voice Calling’ and
  - ’Hunt Group’ the activation and deactivation options are not provided. The display of the current status (On/Off) is also not restricted.
- In the ‘Key Layout’ function the menu items ’Change’ and ’Delete’ are not provided and the allocation of a new key function by dialling a code is prevented. The display of the current key layout is not restricted.
- To prevent entry into the locked branch when the service function is active, pressing the following keys is acknowledged with the temporary display of ’Not Authorised’:
  - Call Forwarding key
  - Direct Station Select key
  - Callback key
  - Reminder key
  - Deputy key
  - Redial key for stored numbers
  - Direct Destination Selecet

- In the Quiet menu, the menu items ’Display Callbacks’ and ’Variab. Forwarding On’ are not provided.
- Pressing the keys listed below, which effect an implicit activation of the service function, is acknowledged with the temporary display of ‘Not Authorised’ when the service function is not activated:
  - Callback key
  - Reminder

Restart

Functions are locked to the same extent as for other static data.

Following a hard restart or a soft restart, the status (function enabled or locked) as last set up by the user is set on the terminal unit.

If user data is reloaded from an external storage medium (Reload), then the status that was set up at the time the data was stored is set.

5.33 One Number Group (ONG)

The feature One Numbering Group (ONG) combines several phones in a one Group. Incoming calls are signaled at more than one destination in parallel (Parallel Ringing Group). For outgoing calls of a Parallel Ringing Group the master extension number is displayed.

A Parallel Ringing Group consists of one master and up to two ONG members (subnumbers). If required a subnumber can be replaced by an external device (e.g. GSM mobile phone). The maximum capacity of an ONG are three members. All members can be reached via a common extension number (master number). ONG internal calls can be established by dialing the internal number (subnumber).

If one of the slave phones in the ONG is out of service, the other phones work as usual. If the master phone in the ONG is out of service, any incoming call to the group is blocked. If one station of the parallel ringing station group is busy, an incoming call is blocked by busy

An incoming call is signaled at every group device. While the call is ringing at the ONG members the calling device sees the ONG master extension number as called number. After answering the call its display does not show the real connected party. Calling line informations are displayed at the master and slave devices. Depending on the type of call (internal or external) the internal ONG members are ringing differently (not possible for the external ONG member i.e. GSM phone). A call to any member of a group is signaled on all member devices as a group call.

When the option SMPF (Search mobile phone first) is activated all incoming basic calls are forwarded to the external device (GSM mobile phone).If a defined time is reached all internal group members are alerted.

Calls between a ONG are only signaled at the respective device, never as a group call.

The following phone types may be a member of an ONG:

– Analog telephone
– digital system telephone
– IP-telephone
– optiClient 130
– CMI cordless phone
– GSM mobile phone

Interaction with other features

Message Waiting Indication (MWI)

MWI is only stored at the master and multiplied to all secondary devices. Each secondary device has access to the master mailbox. A simultaneous menu access is prevented.

Call Forwarding/Call deflection

A call forwarding of the master works as a group feature. Each group member can activate/deactivate the call forwarding. The settings are valid for all group members. A Call Deflection of external calls (no group internal call) is able to the fixed Call Forwarding destination.

Do not disturb

An activated Do Not Disturb feature for the master operates as a group feature. Changes by internal group members are possible.

Call Waiting

When Call Waiting is activated and one of the group members is busy, an incoming call is signaled on all idle members.

Search mobile Phone First

The SMPF feature is only supported in basic call situations (i.e. no support in connections which are on hold)

Call logs

All calls are stored in the call log of the master device. All secondary devices have access to these data.

Call Completion

An activation of Call Completion on no reply by a group member is not available for the group. The activation of CCNR by an incoming call is valid for the complete group.

Recalls

All type of recalls to the group are only signaled on the transferring device

Keyset

Extensions of ONG devices may not appear as key buttons on other keysets

Hunting group

Master devices of an ONG may operate as a member of a hunt group (but not as master). Secondary devices of an ONG must not be a member of a hunting group

Handover

Handover of calls to other extensions between group members is possible (analog and digital). Handover from or to external GSM phone is not supported.

Not supported features

The follwoing features are not supported by ONG:

- CHESE (Chief-Secretary) is not supported for ONG members
- ONG members can not be configured as Night Station
- ONG members can not be configured as ACD agents
- ONG members can not be member of a call pick-up group
- Follow-me is not supported for ONG members
- Emergency override/Emergency intrusion is not supported
- When Call Waiting is activated then SMPF is automatically deactivated
- MLPP is not supported

Restrictions

- The group members of an ONG must be located in the same HiPath node
- ONG is a voice feature only
- An activated Call Forwarding at the external GSM-phone can only be executed in case of correct signaling. As this cannot be guaranteed over all possible signaling paths, it is recommended to keep the GSM phone always active.

5.34 Personal device group

Personal Device Group supports the usage of two devices for one extension number in a HiPath 4000 system. These are usually one desktop phone and one cordless phone (CMI). The Personal Device Group concept is based on keyset functionality to allow multiple devices using the same number.

Keyset configuration is from HiPath 4000 V4 R2 on allowed also for CMI. The CMI device has the Personal Device Group number as prime line (Personal Device Group master) and the desktop phone has the Personal Device Group number as secondary line (Personal Device Group slave). The Prime Line of the Desktop Phone is not used.

Outgoing calls from the cordless or the desktop phone transmit always the same number (Personal Device Group number) to the destination. Incoming calls are signaled on both devices and the user decides where to answer the call.

Supported Devices

- Personal Device Group master: CMI Cordless (S2, SL2, S3-SL3)
- Personal Device Group slave
- optiset TDM
- optiPoint TDM

– OpenStage TDM
– optipoint HFA (IP)
– OpenStage HFA (IP)

Comparison of functions between Personal Device Group and One Number Group:

Required ONS functionality	Personal Device Group	ONS / Parallel Ringing
Number of possible terminals per group that can be reached under a single number (incoming and outgoing)	Master: 1x Cordless Slave: 1x Systemphone	Master: 1x Systemphone Slave: 2x Systemphone + 1x external phone like a mobile phone 1
Inclusion of an external terminal	No	Yes (e.g. mobile)
Busy signaling	Yes	Yes
Same master number in idle display on master and slave	Yes	No
Monitoring for group	Yes	Master subscriber only
CTI for group	Yes	Master subscriber only
DSS key from group	Yes	Yes
Team Executive	Yes	No
Chese	No	No
Synchronized missed calls list	Yes, in combination with DTB	Yes
Support from DTB and BLF-WIN applications	Yes (current versions required)	No
Integration in pickup group	Yes	No
Callback busy/no reply from group	Callback message to all group members	Callback message to group member where callback was activated.
MWI	At group master	Yes
Call Waiting	Yes	Yes
Do Not Disturb	Yes	Yes
Call Divert	Yes	Yes
Ringer Cut-Off	Can be activated for master and slave group member	No

Restrictions

– When parameter TWINNING is changed from YES to NO, the extension number on the phone displays are updated only when the phone idle display is shown / updated

- At the same time only one of the devices can initiate a call with Personal Device Group number.
- If the CMI is talking the desktop phone can not initiate a call with Personal Device Group number
- The master device (CMI) has exactly 1 line (its own prime line) configured.
- The slave device has exactly 2 lines configured (its own prime line and the prime line of the CMI as a secondary line).
- Although the CMI is used as a keyset there are no line keys supported for that device.
- There is only one CMI allowed for a Personal Device Group group.
- Attendant is not supported by the Personal Device Group feature.
- All members of the Personal Device Group configuration must be assigned to the same WABE group.
- All members must be in the same switch. It is not possible to combine a desktop phone of one switch with a CMI configured in a different switch.
- Some menu items are handled different on desktop phone and CMI.
- One device can only be in one Personal Device Group group.
- No other keysets may be combined with the Personal Device Group group.
- The local Call Log functionality does not support Personal Device Group. It must not be used.
- Boss/secretary functionality ist not supported

Interaction with other features

1. Incoming call:

In case of an incoming call (intern or extern) to Personal Device Group both devices the desktop phone and the cordless device ring simultaneously. If one of these devices answers the call the other device stops ringing.

2. Outgoing call:

In case of outgoing call (intern or extern) from Personal Device Group the Personal Device Group number will be displayed on the called device as the calling number no matter if the call was initiated from the cordless or from the desktop phone. This behaviour is also the same in case of call forwarding type unconditional, busy and no answer.

3. User interface:

On both devices of the Personal Device Group group the Personal Device Group number is displayed as the extension number belonging to the device. In general the display structure is different in case of cordless and desktop (e.g. optiPoint) phones. This is also the case for the members of a Personal Device Group.

4. Callback busy / no answer

Callback is supported for a Personal Device Group for internal endpoints only. The callback will be signaled and can be executed on both endpoints of the Personal Device Group. External devices will not be supported for callback. Therefore no callback busy is possible on external devices.

5. Call Forwarding

Call forwarding can be configured on any device of the Personal Device Group. The feature is activated for both devices and the signaling (Display, LED) is done on both devices. Call Forwarding can be activated for the Personal Device Group number using the following ways:

- Cordless / desktop device menu
- AMO ACTDA / AMO ZIEL for the Personal Device Group number
- ACL Feature ACT Deact Service on CTI interface

Call Forwarding will be activated for the Personal Device Group number: If a device calls the Personal Device Group number the call will be forwarded to the forwarding destination and none of the Personal Device Group devices (desktop or cordless) will ring. If the Personal Device Group device has a Call Forward busy activated then the call is forwarded to the CFW destination, e.g. digite device, Attendant, Voice Mail in case of the Personal Device Group number is busy. Therefore either the cordless device is busy or the desktop phone is busy using the Personal Device Group number. Call Forwarding can be activated/ deactivated on any of the Personal Device Group devices.

6. Camp On

Automatic camp on can be activated /deactivated on any of the Personal Device Group devices. Manual Camp On (caller activates camp on) is not supported. Automatic Camp On has to be activated on both devices (Personal Device Group master and slave). This is automatically done synchronously if one of the devices activates or deactivates the feature.

If an incoming second call is calling the Personal Device Group it will camp on the active device only (the one using actually the Personal Device Group line). Camp on is not signaled on the second device of the Personal Device Group.

7. Call Pickup

The Personal Device Group number can be configured to be a member of a call pickup group. The incoming calls to another device in the call pickup group can be picked up by any member of the Personal Device Group (cordless and desktop phone). The incoming calls to the Personal Device Group number can be picked up by any other member of the call pickup group.

8. Call features during an active call

The following call features are supported by the cordless device during an active call: Consultation, Transfer, Toggle and Conference

9. Park / Hold

An active call (that is in call state) can be put on hold the following ways:

– by selecting Halten/Hold in the phone menu (cordless phone, desktop phone).

– by pressing the line key (desktop phone).

This call can be retrieved by the desktop phone by pressing the blinking line key belonging to the phone.

The cordless phone cannot retrieve a call that has been put on hold this way. If the timer has expired the Personal Device Group will be ring back and both member of Personal Device Group can accept the call.

10. Hunt groups

Personal Device Group can be a member of the hunt group. When the hunt group routes an incoming call to the Personal Device Group number, both Personal Device Group devices start ringing and any of the Personal Device Group devices can answer the call.

11. Do not disturb

Do not disturb can be activated/deactivated on any device of the Personal Device Group. The signaling (based on device possibilities) is done on both devices of the Personal Device Group.

12. Timed Reminder

Timed Reminder is supported with Personal Device Group.

13. Speed Dial List (SPDI)

Speed dial list can be used on both devices of the Personal Device Group. Both devices use the same short calling list, i.e. changing the short challing list on one of the devices will be visible also on the other device.

14. Ringer Cut Off

For ringer cutoff two keys can be configured on the desktop phone of the Personal Device Group: ringer cutoff for desktop phone and ringer cutoff for CMI phone.

15. Desktop phone

When ringer cutoff is activated for the desktop phone, incoming calls to the Personal Device Group number are indicated in the following way:

- On the cordless phone the call is indicated as if ringer cutoff was not active
- On the desktop phone there is no ringing / display update. Only the blinking line key indicates the call.

16. Cordless phone

When ringer cutoff is activated for the cordless phone, incoming calls to the Personal Device Group number are indicated in the following way:

- On the cordless phone the call is absolutely not indicated. Note that during such a call attempt the cordless phone seems "idle" however it is not possible to initiate outgoing calls using this device.
- On the desktop phone the call is indicated as if ringer cutoff was not active.

17. Direct Station Selection (DSS) Key

If one of the devices has a configured DSS Key with configured destination of the Personal Device Group on the DSS key the busy condition of the Personal Device Group is displayed via a LED. If the Personal Device Group is called despite being busy, call waiting is automatically activated. By pressing the flashing DSS key while the Personal Device Group is being called, it is possible to take the call if the calling party and the Personal Device Group are in the same pickup group.

A DSS key can only be configured on the desktop phone of a Personal Device Group.

18. Class of Service Switchover (BERUM)

The class of service switchover feature can be activated and deactivated either on the desktop or on the cordless phone. The desktop and the cordless phone must be in the same COS group.

19. Keyset

A Personal Device Group must not be combined with other Keysets. The CMI is only allowed to have one line configured and for the desktop phone exactly two lines are required (own prime line and Personal Device Group number as secondary line).

20. CDRE

CDRE is supported for devices of the Personal Device Group. In the accounting tickets the Personal Device Group number (twincmi) will be reported as paying party.

21. SPE

SPE is supported for devices of a Personal Device Group.

22. ACL Monitoring

In monitor events always the Personal Device Group extension number (twincmi) is reported.

23. ACL Services

ACL services which support the Personal Device Group number:

- Service Comment
- Alternate-Call-Request
- Clear-Connection-Request
- Callback-Request
- Cancel-Callback-Request
- Conference-Call-Request
- Consult-Call-Request
- Deflect-Call-Request
- Dial-Digits-Request
- Divert-Call-Request
- Feature-act/deact-Request
- Feature-Status-Request
- Generate-Digits-Request
- Generate-Telephony-Tone-Request
- Make-Call-Request:
  Restriction: Not working for a Personal Device Group.
- Make-Predictive-Call-Request
- Reconnect-Call-Request
- Single-Step-Transfer-Call-Request
- Transfer-Call-Request
- Device Type Check CMI is reported for the Personal Device Group number.
- Get-Switching-Funct-Devices:
  New device type "Personal Device Group" is reported for twinning number.

– Monitor Set/Cancel
– Snapshot-monitor
– Snapshot-call
– I/O services:
  The Personal Device Group number is to be used to access the CMI. Primary line extension number of desktop phone is to be used to access the desktop phone.
– Set-Lamp-Mode:
  The Personal Device Group number is to be used to access the CMI. Primary line extension number of desktop phone is to be used to access the desktop phone.
– Answer-call-Request It is working only for the desktop phone.
– Hold-Call-Request Service is disabled.
– Retrieve-Call-Request Service is disabled.
– Invoke-Feature Service is disabled.
– Button-Press-Request Service is disabled.
– Escape services Service is disabled.
– ACL Make Call Service is not working for a Personal Device Group.

6 Attendant Console

6.1 Features

6.1.1 Extending Trunk Calls

1. Extending Incoming Trunk Calls Without Announcement (Speed Servicing)

- Extending if the desired station terminal is free
  - During dialling the attendant is still connected to the calling user.
  - The attendant terminal becomes free when the last digit has been keyed. Ringing tone is sent to the desired station terminal; the waiting user is through-connected.
  - Speed servicing is also possible after recall.
- Extending if the desired station terminal is busy
  - The attendant remains connected to the calling user until she withdraws from the connection.
  - Speed servicing is only possible for connections on call keys and additionally the Loop keys from 1 to 6 .
- Retesting a busy station terminal
  If the station terminal becomes free before the attendant withdraws from the connection the desired user is called, provided there is no other call waiting for this user.
  - The display changes to <FREE>.
  - Busy tone to the attendant changes to ringback one if the ATND is connected to the switching side.
- After the attendant changes to the switching side or withdraws from the connection, the waiting user receives music or the "Please hold the line" announcement until the desired user or, in the case of a recall, the attendant answers.
- Preventing the extension of calls for which no connection is possible
  - In the case of speed servicing to a free station which does not have the class-of-service appropriate to the call, automatic switching does not take place; instead, the attendant remains connected to the calling user.

- The station is called when the AS key is pressed. The attendant is then on the station side (V LED lights) and hears the ringback tone. She can now talk with the user (once he has answered).
- Pressing the S key - during conversation with either the calling user or the called user - does not lead to through-connection but causes the external connection to be held on the A key and the station side to be cleared down.

- Non-extendable calls
  - Terminal not equipped
  - Station line circuit does not exist
  - Call number or abbreviated call number does not exist
  - Station line circuit out of operation
  - All trunks busy
  - Invalid calls for station group division
  - No destination stored (speed calling, repertory key)
  - The station does not have the class-of-service appropriate to the call
  - The station has do-not-disturb feature (except if DND override is possible)
  - There is a call already waiting for the desired user (waiting calls can be either extendable or non-extendable, depending on the system setting)
  - After break-in in the event of WA/attendant intercept (a call can be extended when the internal user has answered or before answering, depending on the system setting).
- Extending of trunk calls
  - to an internal user,
    - If the user to whom a call is to be extended has activated call forwarding - all calls, the call will be extended to the forwarding destination.
  - to another attendant terminal,
  - to a user in the satellite PABX,
  - to the Voice Mail Server ,
  - Speed servicing is also possible to external destinations if the line offers "detected end-of-selection/free".
- Speed servicing of trunk calls via lines without end-of-selection is not possible.

- The attendant must through-connect the call manually with the S key or another call key.
  At any time before through-connection the attendant can press the V key to go over to the station side in order, for example, to receive the dial tone (free or busy).
- In the event of ISDN connections, the ATND enters the "Talk" state during exchange signalling "End of dialling, subscriber idle". This makes it possible to extend calls via the V-key even without the answering code when the internal user has answered (it is thus possible to extend announcements from the exchange which are connected without answering code because it is a no-charge call).
- After through-connection of a trunk call to a station in the satellite PABX, the call key used for switching is not released immediately. The LED of the call key is extinguished. When the information on the class-of-service of the satellite PABX user has been received, the call key is released (direct-access user) or it changes to the "automatic hold" condition (no-access user).
  In the hold condition the LED of the call key flashes slowly. If the call key has not yet been released, any attempt to seize it is ignored (R key).

2. Attendant Camp-on - (Exchange) Trunk Calls

- When extending an incoming or outgoing trunk call to a busy station terminal the attendant can withdraw from the call before the station terminal becomes free (retesting for idle condition).
- Attendant release function for ongoing calls only

  With the appropriate AMO, you can determine whether the attendant terminal's release key function applies to

  – ongoing calls and waiting calls only, except waiting exchange calls, or

  – also to waiting exchange calls.
- When the station terminal becomes free the call is automatically extended.

  When the desired station terminal goes on-hook it receives the ringing tone. When the user answers he is connected to the waiting user.
- Only one external or internal call may wait in the queuing memory at any one time for a station terminal to become free.
- When the user with a digital system- and/or IP-telephone clears down his initial call by pressing the TR key, he receives dial tone and can establish a new call.
  The waiting call is not signalled until the Digite user goes on-hook.
- Attendant camp-on for busy hunting group

- The maximum number of instances of attendant camp-on per hunting group is set for each individual hunting group with the aid of administration and maintenance.
- Notification for hunting group users if there is at least one call waiting for the hunting group
  - pulse dialling/pushbutton dialling: No notification
  - digital system- or IP-telephone: Visual call waiting indication
- Waiting calls are extended in their order of arrival to the next hunting group station terminal to become free.

- Notification to the attendant that there is already a call waiting for the desired user.
  - At the end of dialling <BUSY INTERNAL> or <BUSY EXTERNAL> or <BUSY/CALL WAITING/NO CONNECTION POSSIBLE> is displayed.
  - Notification for hunting groups if there is a call already waiting for each hunting group user.
- The waiting user receives music or the "Hold the line please" announcement (options) until the desired user or the attendant answers (in the case of a recall), or no acoustic notification.

  As an alternative to the standard music options, the waiting party can be supplied with a melody or announcement selected by the customer. A separate device port is provided for this purpose via SLMA.

3. Extending Outgoing Trunk Calls

- A user who wants a call to be set up by the attendant calls the attendant by dialling the answering code.
  After notifying the call request the user goes on-hook. The attendant then sets up the desired connection and extends it to the user.
- Extending an undialled trunk
  - The attendant seizes a trunk and extends it to the user. The user can then dial the number he requires.
  - An extended undialled external trunk which is not used by the user within a certain period of time is cleared down when this timeout period expires.
  - As of HiPath 4000, undialled lines can be extended network-wide, provided that the HiPath feature "Least Cost Routing" is implemented in all nodes of the network.
  - Undialled lines cannot be extended to busy stations, except

    to users with digital system- or IP-telephone, if they have activated the "permit camp-on" features. The users in question receive a visual indication (display) and an audible notification (alerting tone).

- Extending a partially dialled trunk is not possible.
- Class-of-service changeover for the user on extending an undialled trunk
  - For the time that the connection is extended the user automatically receives a certain class-of-service which is usually higher than his normal class-of-service. Trunks cannot be extended to no-access users.
  - This higher class-of-service is defined with the aid of administration and maintenance uniformly for the system.
- Extending a fully dialled trunk

  The attendant sets up the desired connection and extends it to the user.
- Extending a trunk when call forwarding - all calls has been activated

  When a trunk is extended, call forwarding - all calls that may have been activated for the user to whom the trunk is to be extended is ignored if extension is on a local basis (ATND and user in the same node).
  If extension is on a network-wide basis (CorNet NQ), i.e. if the ATND and user are in different nodes, call forwarding - all calls will not be ignored. An active call forwarding - all calls feature will be ignored throughout the network.
- Extending dialled outgoing tie lines
- With ISDN connections, the attendant console already switches on trunk signalling "end-of-selection, user idle" to a status in which the attendant can extend calls by way of the V key even without an answer signal when the internal user has answered (it is thus possible to extend announcements from the exchange that are applied without the "answer" code because no charge is made).
- After the trunk code or the trunk call number has been dialled with the aid of the pushbutton set, trunk group key, repertory key, speed calling or number redial facility, the attendant can press the GE key at any time before withdrawing from the call to indicate that a charge call is to be made to the attendant terminal when the user has finished his call.
  - The administration and maintenance system is used to define whether pressing the GE key means "with charge call" or "without charge call".
  - For all calls for which the GE key is not pressed the opposite of the function defined for this key automatic applies.
  - The GE LED lights up when the GE key is pressed and remains on until the attendant withdraws from the connection or until the GE function is cancelled manually by pressing the GE key again.

4. Attendant Loop Transfer

- Transfer of incoming and outgoing trunks to another attendant terminal.

- Transfer of incoming and outgoing internal calls, satellite PABX connections and tie trunks to another attendant terminal.
- Transfer by extending with or without announcement

  Each attendant terminal can be selectively accessed by the users and the other attendants via an individual code.

6.1.2 Announcing Trunk Calls

1. Extending Incoming Trunk Calls With Announcement

- A trunk call answered at the attendant console can be extended to the desired station regardless of whether the station is free or busy.
- Extending if the desired station is free
- Extending if the desired station is busy
- The calling user cannot listen in on the announcement to the desired user.
- Retesting if the station terminal is busy
  - If the station terminal becomes free before the attendant withdraws from the call the desired user is called.
  - The desired user is connected to the attendant when he goes off-hook, provided that there is no other call waiting for this user.
  - The display changes to <FREE>.
  - Busy tone to the attendant changes to ringback tone.
- Extending if the desired station has activated the do-not-disturb feature
- Extending to the voice mail system (VMS)

  Forwarding to the VMS
- Preventing the extension of calls for which no connection is possible
  - Automatic holding of the call to be extended on the A key after an attempt to extend the call.
  - The attendant is no longer in the call status (also no dial tone); she can return to the trunk caller after pressing the A key.
  - The call to the desired user is cleared down.
- Non-extendable calls (extending during announcement)
  - Desired station terminal does not have the class-of-service appropriate to the call to be extended.
  - Invalid call for station group division.

- Extending trunk calls
  - to an internal user,
  - to another attendant terminal,
  - to a user in the satellite PABX,
  - to the voice information server only via call forwarding,
- Attendant release function for ongoing calls only
  With the appropriate AMO, you can determine whether the attendant terminal's release key function applies to
  - ongoing calls and waiting calls only, except waiting exchange calls, or
  - also to waiting exchange calls.

2. Attendant Busy Override for Extending Trunk Calls

- The attendant can override during a call to a busy station
  - Override to station
  - Override to the personal line of an attendant terminal
- Override takes place together with testing

  If the desired user has released his line the call is automatically through-connected (if the attendant does not clear down the connection in the meantime).
  The override condition is thus terminated.
- During an override the desired user "A", the caller B, the overriding attendant, and the override tone are connected together (conference ports are used for this purpose).
  The two users and the attendant can hear one another speak to one another.
  - The conference ports are used jointly for add-on conferences, override, and camp-on.
  - The LTG of the overridden station terminal is searched for the three ports needed.
- During the override status both users and the attendant hear the override tone.

  In the case of speed servicing after the AS key has been pressed, automatic changeover to the station side takes place.
- Interrupting the override
  - The override status is terminated if the AS key is pressed during the override.
  - If the AS key is pressed again the original override status is restored.

- Override is only possible for calls in one of the following talking states:
  - standard call state
  - consultation call
  - toggling state (flip-flop)
  - conference state
  - calls dialled from the attendant's telephone

  In the case of hunting groups, override is only possible if all users in the group are busy, and, if configured, the call queue for the hunting group is full. The user whose call is interrupted is the first user in the group whose call is in one of the above states.

  Calls can also be overridden when they are being "camped on" by another user (Call Waiting - Terminating feature, or "knocking"). The camp-on is then terminated (does not affect visual camp-on indication).

  A call which has already been interrupted by means of the override feature cannot be overridden again.

- Station override security
  - if user A is not in the call state (e.g. in the hold, dial or ringing state),
  - for a call in which a user or device with data protection is involved:
    - data telephone
    - paging system
    - public address system
    - dictation equipment
    - recorded announcement service
    - entrance telephone
  - for a call in which override is already in force,
  - for a station if camp-on is in force for its parties,
  - for a user who is himself making use of the override or camp-on facility,
  - for a user who already has two calls (each call can consist of an initial call and a consultation call),

- Signalling for station override security
  - For data protection:
    <PROTECTED> and busy tone (busy tone only if the attendant does not use speed servicing).

  - In all other cases:
<PLEASE REPEAT> and busy tone (busy tone only if the attendant does not use speed servicing).

- In the event of override security (with the exception of data protection) the AS key can be pressed repeatedly until the procedure leads to the destination, i.e. when the reason for override security no longer applies.
- For each station there is a special camp-on facility (queuing memory 1) for overriding by a user or an attendant. This makes it possible to override even for a user for whom there is already a call waiting.

  If, during an override in the course of extending the call, the attendant terminates by withdrawing from the call the held call is extended to this station and assumes the camp-on status (queuing memory 2) if this status is free. If not, the call is automatically held on the call key.

6.1.3 Attendant Recall

1. Attendant Recall - Trunk Calls, Tie-trunk And Internal Calls

- If the user to whom the call has been extended does not answer within a certain preset time, an automatic call is sent to the extending station
  - after extending with announcement,
  - after extending without announcement (speed servicing).
- The timeout period for automatic recall can be set with the aid of administration and maintenance.
- Automatic recall after timeout
  - If the free user does not answer within the timeout period.
    - During automatic recall the attendant terminal and station (also the station in the satellite PABX) are called in parallel.
    - If the attendant or called user answers then ringing at the two station terminals is terminated.
    - The switching side is released if the automatic recall is answered by the attendant.
  - If the called user is to be called again, i.e. has become free after the recall was answered and if parallel ringing is set, attendants simply press the S key in order to connect the caller in this case. If parallel ringing is set, the answering is indicated to both parties as follows:
    - the attendant console receives an appropriate display message

  - the answering party receives a hold notification (announcement or music).

  This feature does not apply for recalls if called user is busy or has become free and parallel ringing is not set.

- – if the busy user does not become free and answer within the timeout period
  - No release of the switching side.
  - After answering the call the attendant can press the S key to extend the call to the desired user or press the AS key to override.
  - If the desired user becomes free while the automatic recall is being answered the display changes to <FREE>.

- Automatic recall at the attendant terminal from which the last switching operation has been initiated

  The automatic recall is signalled at this attendant terminal until the attendant answers or the user on hold goes on-hook.

- If an automatic recall for outgoing external calls is not answered within 90 s, the connection is automatically cleared down.

  The timeout period for automatic release can be changed with the aid of administration and maintenance (default value 90 s).

- Automatic recall in connection with night service
  - – Night service has been activated before the automatic recall occurred by blocking or disconnecting the headset plug:
    The automatic recall is signalled at the night station.
  - – Night service is activated during signalling of an automatic recall at the attendant terminal

- During the automatic recall the waiting user hears
  - – music, or the "Hold the line please" announcement, or nothing.
  - – As an alternative to the standard music options, the waiting party can be supplied with a melody or announcement selected by the customer. A separate device port is provided for this purpose via SLMA.

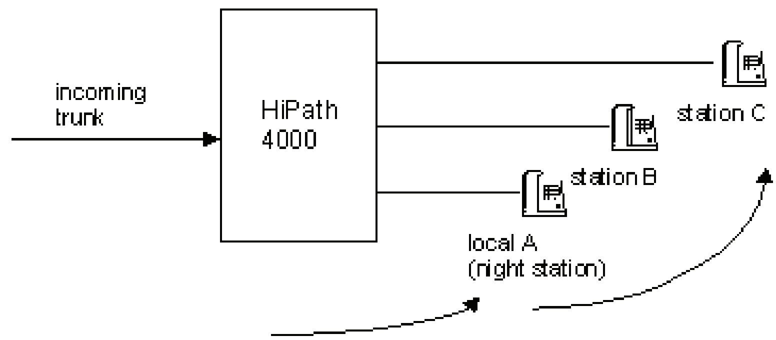
6.1.4 Recalls from Hold

A call to a night station that has been marked as a recall, e.g. recall after unsuccessful call transfer, will follow Call Forwarding No Answer CFNA of the transfering station in the same way as the first incoming call will be diverted to. Normally the recall to a call transfering night station stays at that station and doesn't follow Call Forwarding No Answer CFNA.

The requested recall with following Call Forwarding No Answer CFNA will occur in the following 4 scenarios:

Local night station A with CFNA to station C, station A answers the incoming night call:

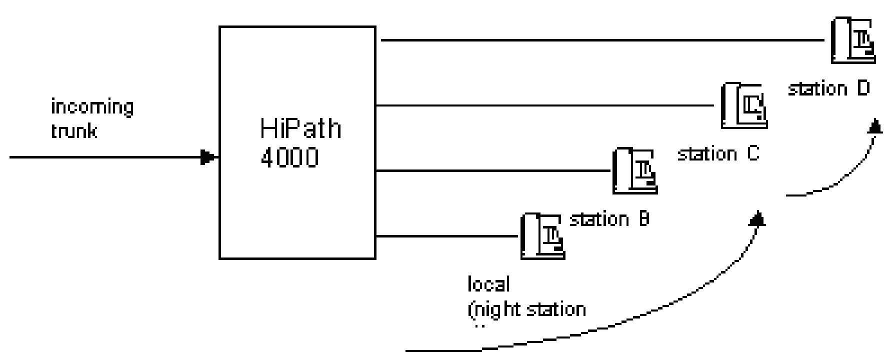
Local night station A transfers the incoming night call to station B. If st. B does not answer it leads to a recall to the local night station A. If st.A does not answer the recall, the recall will be diverted to st. C.



recall to station A with CFNA to station C

Local night station A with CFNA to station C, station C with CFNA to station D:

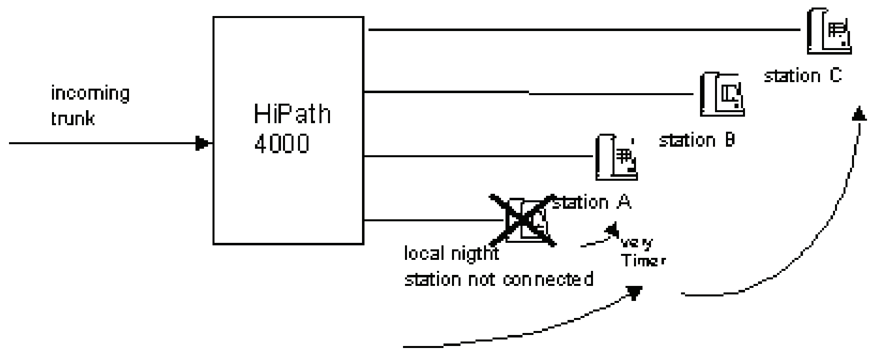
Station A does not answer the incoming night call, the incoming night call is diverted to station C, which answers the call and transfers it to station B. If st. B does not answer it leads to a recall to station C. If st. C does not answer the recall it leads to CFNA to station D.



Recall to station with CFNA to station D

Local or networkwide station A reached via "very short timer" with CFNA to station C, station A answers the incoming night call:

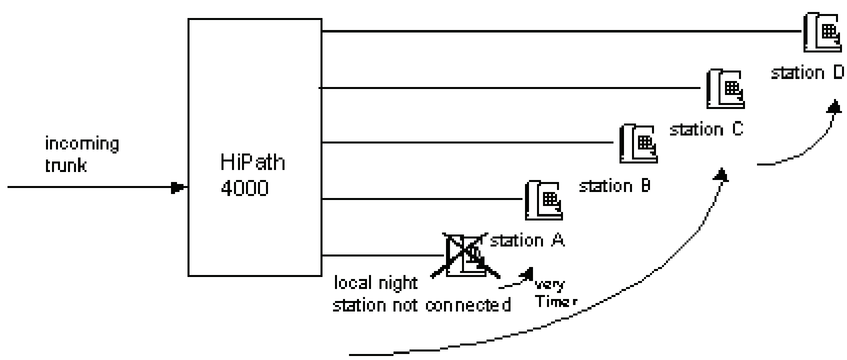
Station A transfers the incoming night call to station B. If st. B does not answer it leads to a recall to station A. If station A does not answer the recall, the recall will be diverted to station C



Recall to station A with CFNA to station C

Local or networkwide station A reached via "very short timer" with CFNA to station C, station C with CFNA to station D:

Station A does not answer the incoming night call, the incoming night call is diverted to station C, which an-swers the call and transfers it to station B. If st. B does not answer it leads to a recall to station C. If st. C does not answer the recall it leads to CFNA to station



Recall to station C with CFNA to station D

Feature Operation

- This feature of a recall to a call transfering night station with following CFNA has to be handled as a system option.
- If this system option is switched off (that is the normal case) the recall to a call transfering night station stays at that station and doesn't follow Call Forwarding No Answer CFNA.
- When there is a call to a night station, which follows CFNA accordingly to this feature the general call forwarding rules of Multiple hop Diversion shall be followed. With the terminology of MHD it means "Call Forwarding Station (variable)" supercedes "Call Forwarding System".

Restrictions

When there is a call to a night station, which is also configured as a master of a Pilot Hunt Group, then the call will not hunt within group. And also the recall will not hunt within the group.

6.1.5 Holding Trunk Calls

1. Attendant Call Hold - Trunk Calls

- Temporary holding of incoming and outgoing trunk calls (automatic hold)
  - Holding during speed servicing (dialling)
  - Holding immediately after pressing the V key
  - Holding if an attempt is made to extend a call for which no connection is possible
  - The information of the held call are displayed in the Loop keys.
- Holding incoming and outgoing trunk calls over a long period
  – Trunk calls can be held so that the attendant terminal is free for further trunk calls during the holding period.
- A user on hold hears music, or the "Hold the line please" announcement, or nothing.
  In the case of digital system- or IP-telephone/attendant terminal, <HOLD THE LINE PLEASE> is also displayed (for a maximum of 5 s).

  As an alternative to the standard music options, the waiting party can be supplied with a melody or announcement selected by the customer. A separate device port is provided for this purpose via SLMA.
- An internal user placed on hold cannot go on consultation hold, but a user of a digital- or IP-Telephone can accept a call in this status.

2. Two-Way Splitting

The attendant can switch between two calls at the attendant terminal any number of times.
There will then always be one call on hold (automatic hold). The user on hold cannot listen in.

6.1.6 Call Park

This feature is implemented via existing AMO and/or other administrative procedures currently used for station and trunk information assignments. This feature has inherent interaction with the following features:

– l Speed Extend - The attendant is able to make use of the Speed Extend capability of the attendant console to direct a call to the park location.

– l System Call Park - Only the 10 system-wide park locations are available, whether calls are parked from stations via the System Call Park feature or by the attendant via the Call Park - Attendant feature.

– l Automatic Recall of Held Calls - The automatic recall feature for calls that have been held in excess of a predetermined time period is also applied to parked calls, and in particular, to calls parked by the attendant as described by this feature.

Feature Access

There are two methods to invoke this feature

1. Automatic assignment of the park location by the system or
2. Specific park location assignment by the attendant

The basic "access code" is used to automatically park the call; the system responds (via the attendant display) with the assigned park slot number, if one is available. Optionally, the attendant are able to park a call in a specific location by dialing the access code plus the digit representing the park slot location.

Remark: Because there are a limited number of park locations available in the system, use of the Automatic Park Location Assignment (described in the following paragraphs) is preferred to that of selecting a specific park location.

Automatic Park Location Assignment / Specific Park Location Assignment:

The attendant is able to invoke this feature either by Access Code (followed by a park location number) or via a Direct Destination Selection (DDS) key which has been preprogrammed with the Access Code (plus a park slot location number).
The attendant console display provides an indication that the call has successfully been parked, including the specific park location ("slot") as assigned by system software.
An Example Display Message for a call parked in slot 5 is called:

PARKED IN SLOT 5

Upon successful completion of activation of the feature, the attendant call state changes from "talk" to "idle," permitting the attendant to process other calls.

All Park Locations in Use:

If the attendant attempts to park a call by the either of the two methods described above, and all park locations are currently in use, a message is displayed to the attendant indicating same, and the attendant receives Reorder Tone as an audible alert to the condition. The message and the audible
alert is presented to the attendant upon pressing the DDS key preprogrammed for the feature, or upon completion of dialing of the Access Code.
The display message is called: ALL PARKSLOTS FULL

Selected Park Location in Use:

If the attendant attempts to park a call by the method "Specific Park Location Assignment", and that park location is currently in use, but another park location is available, the call is automatically parked in the alternate location and the attendant receives a confirmation message indicating where the call has been parked.
Example: The selected Park Location 5 Busy, the Call is Parked in Location 3.
The display message is called: 5 BUSY, PARKED IN 3

Use of Speed Extend and DEST Key from Attendant Console:

If the attendant is in the talk state with a party and dials the Access Code from the console keypad (as is done when speed extending a call to a destination), the call is parked as described above.
Alternately, the attendant is able to first press the DEST (Destination) key (temporarily holding the call) and then dial the Access Code. This method allows provision for the attendant to park an outgoing call.

Parked Call Retrieval:

Once a call has been successfully parked at one of the system park locations by either of the two methods described, the call is able to be retrieved by either a station party or an attendant, by dialing the Access Code followed by the park slot location digit while idle and off-hook. Upon successful retrieval of a parked call, the party and the attendant are connected in the talk state.

Parked Call Abandoned:

If the parked party (station or trunk) hangs up before the attendant or a station user retrieves the call, or before the call times out as a recall, the entry is deleted by the system (if the system can detect the on-hook condition of the parked party) and the park location is made available for another call.
If the attendant attempts retrieval of a parked call that has hung up, the attendant is provided with a message on the console display indicating that the party is no longer available.
The Display Message is called: PARK SLOT EMPTY
It means: The parked party has hung up

Parked Call Times Out:

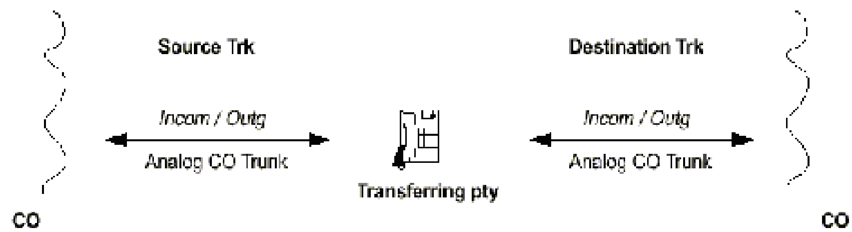
If the parked call is not retrieved by the attendant or a station user within a predetermined period of time, the call is timed out and is recalled to the attendant console in the same manner as for other recalls (e.g., recall of held call).
Parked calls that time out are routed back to the attendant that parked the call or, in the event that attendant is no longer in service, to another attendant. The attendant receiving the recall is provided with a message on the console display indicating that this is a recall (as opposed to a new call).
The Display message is called: **RECALL - HOLD**

6.1.7 AC Supervised Trunk to Trunk Connection

HiPath 4000 provides the possibility to connect two trunks that do not have disconnect supervision and to supervise (release / reextend) this connection

Features

The attendant/station can establish a trunk-to-trunk connection between two analog trunks with no disconnect supervision (trunks that are not able to signalise the release).Such connection is controlled via the trunk-to-trunk recall timer:upon the expiration of the timer , the call is recalled to an attendant console so that an operator can manually supervise the connection.



If the transfer was successful and neither source trunk nor destination trunk has disconnect supervision then, the transferring node starts the automatic trunk-to-trunk recall timer at the time the station leaves the connection.

When the automatic trunk-to-trunk recall timer expires, the trunk-to-trunk connection is recalled to an attendant console so that an attendant can manually check (supervise) the status of the connection.

The connection between the source and destination trunk is kept (they can converse) when the recall alerts the Attendant . At this point the Attendant alerting line informs "RECALL-TRK EXCH" <<trunk group name of the source trunk>> display for the attendant .

Attendant intrusion into the trunks' connection will create "3 party" conference between the Attendant and the Two trunks.

When the recall is answered by the operator and a "3 party conference" is created between the attendant and the trunks audible messages - beep-tone is heard by all involved parties in order to signalize the intrusion of the operator in this call.

At this point if the operator presses the EXTEND key the two trunks are connected again and if the operator presses the RELEASE key the connection is torn down.

The trunk controlled by the timer is shown in the Source Side, the other one in the Destination Side. Both of them with the status "CONNECTED".

If the local attendant group serving the ITR group of the transferring station is in night mode when the recall timer expires, then the night mode timer is started. When the night mode timer expires, the connection is released.

Details about the CP scenarios when trunks with no disconnect supervision may be connected:

A Supervised Connection can be initiated by:

- Analog telephones
- Digital system telephones
- IP-telephones
- Attendant Consoles
- ISDN Trunks (e.g. connection from conference, PhoneMail)

Transfer Talking

A properly classmarked station or the Attendant can establish a trunk-to-trunk connection between two analog trunks and then exit the connection, leaving the connection established so that the two trunk parties can converse.

Two types of analog runk To Trunk connections considered for Transfer Talking:

- Incoming Trunk To Outgoing Trunk
- Outgoing Trunk To Outgoing Trunk

Transfer before answer

A station or an Attendant can perform Transfer Before Answer- Feature. When connecting the trunks one of them has to be answered, the other one being even in dial state.
One type of Analog Trunk To Trunk connections has to be considered for Transfer Before Answer:

Outgoing Trunk To Outgoing Trunk

Transfer from conference

The Transfer From Conference Scenario handles only the situation when two Analog trunks are in conference with a party (can be station or supervised trunk) and this party drops off.

Attendant Join feature

Join extending is used for extending 2 calls on different subunits.
Due to the Join Feature an Attendant can establish a trunk-to-trunk connection between two analog trunks and then exit the connection, leaving the connection established so that the two trunk parties can converse.

Three types of Analog Trunk To Trunk connections have to be considered for Join Feature:

- Incoming Trunk To Outgoing Trunk
- Outgoing Trunk To IncomingTrunk
- Incoming Trunk To Incoming Trunk

Route Optimization (Path Replacement)

When the connection is supervised this feature has to be disabled.

6.1.8 Internal Calls to/from Attendant Consoles

Attendant Calls

- The following types of traffic are treated as attendant calls at the attendant console (initial calls and recalls):
  - Calls from internal users,
  - Calls from users in a network,
- For attendant calls the attendant console is accessed with the aid of the answering code.
- Call distribution comes into force if there is more than one attendant terminal for the attendant calls. An attendant call is signalled at an attendant terminal with the "attendant calls" mode.
- Switched loop operation for attendant calls
- Console answering priorities for attendant calls
  - For internal initial calls with answering code dialling the queue is limited.
  - If the station is busy for an attendant, call override/camp-on is prevented.
  - The limitation can be defined with the aid of administration and maintenance (in the extreme case there is no queuing memory).
- Query of call via pressure of respective source key.
- Attendant calls are placed on hold inthe same way of trunk calls.
- Extending of attendant calls
- Attendant camp-on for extended attendant calls
- Automatic recall for extended attendant calls
- Night service for attendant calls
- If the attendant console is an emergency call station emergency calls (set up with a timed hot-line service) are not rejected if all trunks are busy. Connection setup is automatic repeated on expiry of a timer. This also applies to the night service condition.

Notification Call

Notification is call forwarding to the attendant console. A notification call is a call forwarded to the answering line of the attendant console. Users who activate notification are in "notification" state and are "notification users".

- Notification calls are specially-signalled on the attendant console screen. In addition to the standard call signalling, the name and station number of the notification user are displayed and marked with an arrow, when the attendant answers the call.
- If an attendant calls a user in "notification", the display texts "Notification" and "Not extendable" will initially appear. The attendant must press the AS key in order to override this and ring the user.
- The "notification call" feature functions throughout an integrated network (homogeneous network with CorNet NQ protocol).
  – Condition for network-wide function: The FWD destination of the notification user must be an attendant console group in the user's **own** node. Forwarded calls are then processed according to CAC rules.
- Notification calls can only be forwarded to attendant console groups, and not to individual attendant consoles.
- The notification feature cannot be used in conjunction with the 'forward on no answer' (FNANS) feature, but only with 'call forwarding - all calls' (FWD). If the attendant calls a user who has activated FNANS to the attendant console, the call is **not** forwarded. FNANS from the night answer position to the attendant console is not rejected. FWD from the night answer position to the attendant is always rejected, unless set for the personal telephone number of a night attendant.
- Notification has priority over attendant intercept, do-not-disturb and through-dialling protection.

Personal Calls to Attendant Console

- The attendant can be reached by means of a personal telefon number.
- Personal calls to the attendant console may come from the following station terminals:
  – Internal user,
  – Another attendant (= interposition calling),
  – A user in a network,
  – External users (DID).
- Incoming direct-dialled exchange calls to an unoccupied attendant console are automatically re-routed to another attendant or to the night service terminal, if the absent attendant has unplugged the headset.

- The personal call to the attendant console is placed on hold as in the case of trunk calls
- Extending of personal calls to the attendant console (as in the case of attendant calls)
- Attendant camp-on for extending personal calls to the attendant console (as in the case of attendant calls)
- Automatic recall for extending personal calls to the attendant console (as in the case of attendant calls).

Attendant Busy Override in Internal Calls

- Attendant busy override in internal traffic is the same as attendant busy override for extending trunk calls
- Attendant busy override for extending tie trunk and internal calls
- Attendant busy override for outgoing internal attendant calls (initial call)
- Override is only possible for calls in one of the following talking states:
  - standard call state
  - consultation call
  - toggling state (flip-flop)
  - conference state
  - calls dialled from the attendant's telephone

  In the case of hunting groups, override is only possible if all users in the group are busy, and, if configured, the call queue for the hunting group is full. The user whose call is interrupted is the first user in the group whose call is in one of the above states.

  - Calls can also be overriden when they are being "camped on" by another user (Call Waiting - Terminating feature, or "knocking"). The camp-on is then terminated (does not affect visual camp-on indication).

  A call which has already been interrupted by means of the override feature cannot be overridden again.

Attendant Camp-on - Tie Trunk and Internal Calls

- When extending an incoming or outgoing internal, tie trunk or satellite system call to a busy station terminal the attendant can withdraw from the call before the station terminal becomes free (retesting for free condition).
- When the station terminal becomes free the call is automatically extended.
  - When the desired station terminal goes on-hook it receives the ringing tone.

- When the user answers he is connected to the waiting user.

- Only one external or internal call may wait in the queuing memory at any one time for a station terminal to become free.
- Attendant camp-on for busy hunting group
  - The maximum number of instances of attendant camp-on per hunting group can be set for each individual hunting group with the aid of administration and maintenance.
  - Notification for hunting group users if there is at least one call waiting for the hunting group.
  - Waiting calls are extended in their order of arrival to the next hunting group station terminal to become free.
- Notification to the attendant that there is already a call waiting for the desired user.
- The waiting user receives an announcement until the desired user or the attendant answers.

Extending Outgoing Trunk Calls

- A user who wants a call to be set up by the attendant calls the attendant by dialling the answering code.
  After notifying the call request the user waits with handset off-hook for the attendant to extend the call to him.
- Extending an undialled trunk
  - The attendant seizes a trunk and extends it to the waiting user. The user can then dial the number he requires.
  - Undialled trunks can be extended from the attendant console or the night answer position throughout an integrated network, provided the 'Least Cost Routing' feature is activated in all nodes of the network.
- Extending a partly-dialled trunk is generally not possible
- Class-of-service changeover for the user on extending an undialled trunk
  - For the time that the connection is extended the user automatically receives a certain class-of-service which is normally higher than his normal class-of-service. Trunks cannot be extended to no-access users.
  - this higher class-of-service is defined with the aid of administration and maintenance uniformly for the system.
- Extending a dialled trunk
  - The attendant sets up the desired connection and extends it to the waiting user.

  - If the attendant extends a partially dialled trunk the user cannot dial the missing digits.
- Extending dialled tie lines
- Speed servicing is also possible for extending if the line offers "detected end-of-selection/free".
- Speed servicing for extending to lines without end-of-selection is not possible.
- With ISDN connections, the attendant console switches to the "call" status on trunk signalling "end-of-selection, user idle". The attendant can therefore extend calls by way of the V key even without an answer signal (it is thus possible to extend announcements from the exchange that are applied without the "answer" code because no charge is made).
- After the trunk code or the trunk call number has been dialled with the aid of the pushbutton set, trunk group key, repertory key, speed calling or number redial facility the attendant can press the GE key at any time before withdrawing from the call to indicate that a charge call is to be made to the attendant terminal when the user has finished his call.
  - The administration and maintenance system is used to define whether pressing the GE key means "with charge call" or "without charge call".
  - For all calls for which the GE key is not pressed the opposite of the function defined for this key automatically applies.
  - The GE LED lights up when the GE key is pressed and remains on until the attendant withdraws from the connection or until the GE function is cancelled manually by pressing the GE key again.

Outgoing Internal Call Setup by the Attendant

- The attendant can set up outgoing calls to internal users. The following features are possible here:
  - Speed calling
  - Repertory keys
  - Override
  - Paging
  - Number redial
- Retesting for a busy internal station terminal
  - If the station terminal becomes free before the attendant with draws from the call, the attendant hears the ringback tone and ringing tone is sent to the station terminal.

Interposition Calling

- In systems with more than one attendant terminal the attendants can call one another.
- Each attendant terminal can be accessed via an individual call number by the users and the other attendants.
- In interposition calling the following features are possible:
  - Override
  - Extending (attendant loop transfer) with or without announcement.

6.1.9 Call Distribution and Console Answering Priorities

The attendant groups can be configured either with double call queue or multiple call queue. This choice determinates the call distribution, answering priorities and signalisation of the attendant group.

The HiPath 4000 supports attendant groups with double call queues (AC4, AC-Win 4.0).

1. Call Distribution

- In systems with more than one attendant terminal calls which do not have to be signalled at a particular attendant terminal are circulated among attendant terminals so that the load is distributed among all the attendant terminals as evenly as possible.
- A selective call to an attendant terminal which is out of operation is extended to any other attendant terminal.
- Automatic recall is signalled at the attendant terminal from which the last switching operation has been initiated (selective call)
- Extending of calls to attendant groups
  - Attendant groups are defined with the aid of the following parameters:
    - Code in incoming external traffic
    - Code in incoming internal traffic
    - User group from which the internal call comes
    - Tie trunk or station line group via which an external call comes after the answering code has been dialled
  - The attendant terminals are assigned to attendant groups with the aid of administration and maintenance

- Calls can be set up from any attendant group to users in any user group. During call extending, however, the connection (caller station) is checked for validity.
- Maximum of 16 attendant groups per system
- A call which arrives at a busy attendant group waits until one of the attendant terminals in the group is free.
- A selective call which cannot be extended or a call for which there is currently no attendant terminal available in the attendant group is signalled at any attendant terminal in the same attendant group.
- An attendant terminal may not belong to more than one attendant group.
- Attendant groups can be selectively accessed by external users with the aid of codes or on the basis of trunk groups (applies to trunk and tie trunk calls).
- Flexible assignment of the attendant group code in external traffic to an attendant group enables flexible assignment of the attendant groups to the customers of a shared system.
- The customer name is also displayed on the attendant display during calls.
- In cases in which the attendant group responsible for an incoming external call cannot be established by the system (perhaps because of incomplete dialling) the call is diverted after a timeout period to any attendant group specified for this purpose with the aid of administration and maintenance.
- Assignment of the attendant group to an answering code with the aid of administration and maintenance. In shared systems the administration and maintenance facility is used to define which attendant group is to be accessed for each customer.

2. Console Answering Priorities

- Non-selective calls at the attendant console which temporarily cannot be signalled because all the appropriate call keys at the attendant terminal are in use are camped on.
  - The caller receives ringing tone.
  - As soon as the answering unit at a console becomes idle, the call is signalled.
  - Distribution of the calls among idle consoles according to the type of call is the standard setting. This form of allocation can be controlled (by way of AMOs) in such a way that a call is not signalled at a console until the latter is completely free, i.e. all answering units are idle (including those to which the incoming call would not normally be routed).

- The waiting calls are signalled in the order of their arrival.
- Console answering priorities according to type of call takes precedence over console answering priorities based on order of arrival.
  Calls with the same priority are signalled in the order of their arrival, however.
- Types of traffic and calls with console answering priorities according to order of arrival
  - Trunk calls
  - Tie trunk calls
  - Attendant calls
  - Initial calls
  - Automatic recalls
- If the capacity of the call queuer is exceeded any further call is held until space becomes available in the call queuer.
- For internal initial calls with dialling of the answering code the queue is limited. Calls which arrive at a full queue receive the busy tone.
  - The maximum number of initial attendant calls can be defined for each attendant group with the aid of administration and maintenance
    - For each attendant group a factor of 0 to 5 (initial attendant calls per attendant terminal) can be entered.
    - Limitation of the call queuer is dynamic.
      The system multiplies the number of operational attendant terminals per attendant group by the factor entered for this group to determine the number of call queuer queuing memories for attendant calls.
    - The necessary queuing memories are taken from a pool shared by all the attendant groups.
  - the maximum number of waiting initial attendant calls covers only those calls which have not yet been signalled or are in the call state.
- Flexible customer-specific definition of the sequence of priorities for types of traffic and types of call for call signalling at the attendant console or special night answer position.
- There are 5 priority levels with the following types of traffic and call (the sequence of priorities can be changed)
  :

1.	Emergency call:	– Off-hook recall (timed hotline service)
2.	Trunk calls	– automatic recall
	Trunk call	– serial call
	Trunk calls	– call transfer - attendant

	Trunk calls	–	cradling during consultation
3.	Trunk calls	–	initial call
	Trunk calls	–	attendant intercept
	Trunk calls	–	notification call
4.	Tie line calls	–	automatic recall
	Tie line calls	–	serial call
	Tie line calls	–	call transfer - attendant
	Tie line calls	–	cradling during consultation
	Tie line calls	–	preferred trunk group call
	Attendant calls	–	preferred user call
	Hotline	–	automatic connection setup
	Radio call		
5.	Tie line calls	–	initial call
	Tie line calls	–	attendant intercept
	Tie line calls	–	notification call
	Attendant calls	–	initial call
	Attendant calls	–	automatic recall
	Attendant calls	–	notification call

- Default sequence of priorities for the attendant terminal
  1. Emergency call
  2. Trunk calls / automatic recall
  3. Trunk calls / initial call, notification calls (external)
  4. Tie trunk calls / preferred trunk groups, radio call, call from a preferred user
  5. Tie trunk calls / initial call and automatic recall, attendant calls, notification calls (internal)
- Definition of a customer-specific sequence of 5 priority levels with the aid of administration and maintenance

  The traffic and call types per priority level are fixed.
- The calls are also answered in the order of their arrival within each priority level.
- The console answering priorities apply only to calls in the queuing memory.
- Interposition calls are not subject to console answering priorities.

- The call queuer for each attendant group for attendant calls has a memory location which is reserved for the timed hot-line service. This memory location has priority 1, i.e. even if the call queuer is full this call is the next to be extended to a free attendant terminal.
- Attendant console traffic can be temporarily diverted in the event of an overflow:
  – Incoming calls to other attendant consoles or to the night answering station can be diverted within the node or throughout the network as follows, in order to reduce the load on the given attendant console or night answering station:
    - to the attendant console group with the highest number
    - to an attendant console group defined using AMOs.
  – A full queue is currently the criterion for this forwarding function.

6.1.10 Number Redial – Attendant

- The last external or internal call number dialled with the pushbutton set, speed dialling, repertory key or direct station selection key can be stored for later number redial provided the connection is still established. The number actually dialled is stored and no longer the long number converted from entered speed dialling numbers.
- Maximum number of digits for the saved number redial destination:
  – External destinations:22 digits incl. trunk group code
  – Internal destinations: 6 digits
- All dialling information entered via the pushbutton set, i.e. including * and #, can be stored
- Storing dialling information for an outgoing call
  – Storage of dialling information accompanies speech.
  – Only one call number can be stored. It remains stored until overwritten by another number.
  – The call number to be stored is a call number dialled on any call key in the course of connection setup to the source or destination.
  – The call number just dialled is stored when the WW key is pressed after end-of-selection (in the free, busy or call state or after artificial end-of-selection).
  – The call number stored is the one dialled on the side on which the attendant is located at the time the WW key is pressed.

- Storing dialling information for an incoming call (notebook)
  - In the call state for an incoming call the attendant can enter a call number and store it for later number redial.
    - Storage of dialling information for an incoming internal call
    - Storage of dialling information for an incoming satellite PABX or tie trunk call
  - In addition to the call number dialled by the attendant, the caller's call number is automatically stored. This means that manual dialling is not necessary if the outgoing call then has to be extended.
  - The dialled call number is stored successively as it is entered.
  - The call connection remains intact during and after storage of the dialling information.
  - The dialling information can be cleared by pressing the CL key.
  - A maximum of 10 call number pairs (internal and external call numbers) can be stored.
  - After an unsuccessful attempt to establish a connection by extending (for example because the desired destination is busy), a notebook entry can be made in which the source and destination are stored.
- Interrogating the contents of the notebook
  - Interrogating the list containing a maximum of 10 number redial entries
  - For each entry the list shows the stored call numbers (including the trunk group code for external call numbers) for the source and destination. The two call numbers are shown on the same line.
- Deleting stored destination information
  - Manual deletion
  - Automatic deletion after through-connecting the call set up with the number redial facility to the job originator
- Number redial
  - A stored call number can be re-transmitted any number of times on any call key on the source or destination side; it remains stored until overwritten by another number.
  - In the digit recording state on the originating side
  - In the digit recording state on the destination side
- Number redial with the aid of the notebook

- A stored call number can be re-transmitted any number of times on any call key on the source or destination side; it remains stored until the call is switched through to the job originator.
- Connection setup is possible only with "extending", i.e. the call number of the desired destination can only be transmitted on the job originator side.
- In the idle state
- In the digit recording state on the originating side
- After the line number has been selected and the WW key has been pressed the display changes from the programming layout to the call switching layout.
- After each unsuccessful attempt a fresh attempt must be made to transmit the stored call number.
- If an attempt is made at the attendant terminal in the idle state after interrogation of the WW entries to move the cursor to a line which does not contain stored dialling information, the attendant receives the negative acknowledgment <NOT POSSIBLE>.

6.1.11 Switched Loop Operation

- Switched loop operation for
  - trunks
  - tie trunks
  - attendant calls
  - charge calls
- A call can be signalled on all call keys of an attendant console simultaneously. Waiting calls can be individually processed in any order.
- The busy call key is immediately released when the call is extended; it remains busy, however, if the call is placed on "automatic hold".
- If an attempt is made to extend a call for which no connection is possible, this call is placed in the hold state – irrespective of whether speed servicing or the V key is used – with hold signalling.
- The calls keys are used for initial calls and recalls.

6.1.12 Transfer of Ringing (Attendant Console)

- An external or internal call is automatically forwarded if it is not answered within a certain predefinable time at the attendant terminal at which it is signalled, provided control operations are not being performed at this attendant terminal.
  This also applies to the night attendant console.
- The timeout period for call forwarding is defined with the aid of administration and maintenance.
- With the aid of administration and maintenance, the transfer of ringing facility can be deactivated individually for each attendant group.
- If a call is not answered within the transfer or ringing timeout period on the source keys and there is no call entering state in force on any of the source keys, the attendant terminal is automatically pre-blocked.
- In the pre-blocking state existing calls source keys are forwarded to other operational attendant terminals . If there are none, calls on A and M are forwarded to the transfer station.
- No transfer or ringing for personal calls and self-initiated calls such as callback and timed reminders.
- If there is a call on P and/or there are calls on hold/parked, a timeout period is started at the beginning of the pre-blocking state (only AC4 and AC-Win 3.0).
- The following alternative option can be set with the aid of administration and maintenance:The last attendant terminal in an attendant group is not pre-blocked after timeout.
- Destination for transfer of ringing
  – If there is at least one other attendant terminal which is operational, the call goes back to the call queuer and is extended to another attendant terminal.
  – If there is no other attendant terminal which is operational the call goes to a particular station terminal (transfer station).
- Transfer of ringing to a station terminal (transfer station)
  – The transfer station is entered with the aid of administration and maintenance. Transfer or ringing is then active.
  – After manual activation (night service) the destination for transfer of ringing can also be used as the night station.
  – Signalling at the transfer station after automatic activation of night service
    - Alerting tone if the transfer station is busy

- Type of transfer station
  - normal station
  - night attendant terminal
  - group of stations as the special night answer position
  - universal night answer
- Console answering priorities as for night service
- Transfer of ringing activated at the transfer station is honoured.
- The transfer station automatically receives all necessary classes-of-service, as in the case of the night station.
- There may be no more than one transfer station per attendant group.

6.1.13 Outgoing Calls

1. Outgoing External Call Setup by the Attendant

- The attendant can set up outgoing calls to external users
  - via trunks
  - via tie trunks
  - via the line to the satellite PABX.
- In setting up an outgoing call the following services are possible:
  - Speed calling
  - Repertory keys
  - Trunk queuing
  - Number redial
- The call is released by pressing the TR key.
- Attendant release function for ongoing calls only
  With the appropriate AMO, you can determine whether the attendant terminal's release key function applies to
  - ongoing calls and waiting calls only, except waiting exchange calls, or
  - also to waiting exchange calls.
- For lines without an end-of-selection criterion the system switches to the call state if a digit is not dialled within 7 seconds of the previous digit.

- With the aid of administration and maintenance it is possible to define whether after seizure of the trunk dial tone is to be sent or not. The trunk dial tone is generated by the system. It arrives irrespective of whether the exchange is ready for dialling or not.
- The second trunk dial tone and other trunk dial tones are evaluated by the dial tone receiver but are not sent to the attendant. A maximum of 6 dial tone evaluations can take place per dial operation. The spacings (number of digits) are defined with the aid of administration and maintenance.

6.1.14 Automatic Dialler for the Attendant Console (Repertory Keys for the Attendant Console)

- The attendant has the following repertory keys at her disposal:

  Each of the 42 repertory keys can be used for dialling a call number (repertory key) or a trunk group code (trunk group key).
- An internal or external call number, a trunk group code or a code procedure can be stored on each of the repertory keys.
- Maximum of 22 digits per destination (including the trunk group code for external destinations)
- All dialling information entered on the pushbutton set can be stored, (in other words, including * and #).
- It is not possible to enter functions of feature keys (key codes) on the repertory keys.
- In the maximum configuration all the attendants can have repertory keys.
- Manual suffix dialling for external calls
- Entering a destination
  – The attendant can enter and store an internal or external destination, a trunk group code or a procedure on any repertory key with the aid of the pushbutton set.
  – Any call number already stored on the repertory key is overwritten.
  – A number of destinations can be entered one after the other without having to start the procedure from the beginning each time.
- Deleting a destination

  The destinations entered by the attendant can be deleted
  - by overwriting with another destination,
  - with the aid of the delete procedure.

- Interrogating the destinations
  - The attendant can interrogate the call numbers stored on the repertory keys.
  - All the destinations can be interrogated one after the other.

2. Extension before a call

This feature allows attendants to put calls to ISDN subscribers on hold during the call setup period for internal consultation calls, i.e.,

  - outgoing external calls to ISDN subscribers can be placed on HOLD without having to wait for the CONNECT message
  - dialled outgoing external calls to ISDN subscribers can be transferred to internal users regardless of the outcome of the connection setup (provided the connection is set up at one end, i.e. at least one party has answered).

- Standard Procedure
  - The attendant dials an external ISDN number. The attendant can immediately place the connection on hold without having to wait for a response from the ISDN subscriber line and set up an internal (consultation) call to a HiPath 4000 user. The attendant console display shows the standard setup status messages (e.g. ringing or connect messages) received from the ISDN subscriber line after the connection has been placed on hold.
  - Once the internal HiPath 4000 call has been initiated, the attendant can transfer the call, provided at least one of the two parties answers (i.e. CONNECT message must be received from external ISDN line OR internal user). If the second party does not answer, the first party can only disconnect, i.e., other features such as callback etc. cannot be used.
- Restrictions in Conjunction with Other Features
  - LCR cheapest route search is cancelled, if the attendant switches over to the HiPath 4000 side before the ISDN subscriber line is reached.
  - Consultation via same (exchange) line is not supported.
  - Functional Services are not supported.
  - Transfer by Pull is not supported
  - Toggling
    - Attendants cannot return to the exchange side (1st call setup) after changing over to the HiPath 4000 side (2nd call setup) before the exchange side answered.

- Internal HiPath 4000 users cannot transmit further dialling information after call transfer, i.e., if the dialled ISDN number is incomplete, it cannot be completed by the internal user.
- Callback requests are not supported (e.g. if the ISDN subscriber does not answer).
- Incomplete ISDN connections cannot be parked.

6.1.15 Variable Night Service Assignment

- Answering, selecting, and activation of night service variants is possible from any attendant terminal (including the night attendant terminal).
- There are a maximum of 8 night service variants per attendant group.
- A shared attendant console only has shared night service variants for all the customers of a shared system.
- Night service variants are entered, changed, and deleted with the aid of administration and maintenance.
- Answering of the night service variants with procedure.
- With the aid of administration and maintenance a night service variant can be entered for which signalling of the calls switched for night service is suppressed.
- Selecting the night service variant
- Before the night service variant is activated the system must be informed about which of the night service variants entered is to become active.
- Activating the night service
  - The last night service variant selected from any attendant terminal can be activated at the last operational attendant terminal.
  - New calls are signalled according to the night service variant selected.
  - Waiting calls are returned to the call memory and then signalled in the same way as a new call according to the night service variant selected.
  - Existing attendant calls are released or through-connected if connections are possible.
  - Calls placed on hold by the attendant are signalled by continuous ringing
  - When the handset/headset is unplugged the activated night service variant is displayed for 5 seconds.
- Deactivating the night service

  – The night service can be deactivated from any attendant terminal.

- Another night service variant of the appropriate attendant group can be selected and activated from the night attendant terminal.

6.1.16 Different Attendant Console Access Codes for Incoming Exchange Trunk Calls

- If trunk calls are to have different attendant console access codes and/or specific attendant terminals or groups of attendant terminals are to be accessed, the digit 0 for accessing the attendant console in incoming central office traffic can be replaced by a digit code .
- Codes are assigned to attendant groups with the aid of administration and maintenance. More than one code may be assigned to an attendant group.
- Codes are evaluated by the system
  – for assignment to the appropriate attendant group
  – for displaying the relevant customer name on the console display (in shared systems).

6.1.17 Trunk Reservation

The attendant can block a certain definable number of external lines for outgoing seizure by internal users. this operation does not seize the trunks. The trunks can then be accessed only by the attendant, external users and certain internal users.

- With the aid of administration and maintenance any external lines can be defined for reservation.
- With the aid of administration and maintenance it is possible to define the users who may access the reserved external lines.
- If the attendant console consists of a number of attendant terminals, trunk reservation can be activated and deactivated from any attendant terminal.
- Reserved lines can be extended or assigned by the attendant and transferred by users with the appropriate class-of-service.

6.1.18 Serial Call

- An external call is signalled at the attendant console as an automatic recall after the internal user to whom the call was extended goes on-hook, provided the attendant pressed the serial call key before through-connecting to the first user.

- Serial calls are possible for
  - incoming calls
  - outgoing calls
  - trunk calls
  - tie trunk calls
- A system or even an attendant terminal may handle more than one serial call at any one time.
- In the night service state the automatic recall is forwarded to the night station.
  - The serial call function is deactivated when the call is answered at the night station (station terminal).
  - The serial call function can be re-activated at the night attendant terminal.
- The automatic recall is signalled at the attendant terminal at which the initial call was answered.
- Charge allocation for outgoing serial calls
  - Each internal user is allocated charges for the proportion of the serial call in which the connection was in the call or hold state at his station.
  - The charges for the call state at the attendant terminal and for extending (waiting state) are allocated to the user to whom the call is being extended.
  - Call charges incurred during the automatic recall are allocated to the attendant or, if the call is extended again, to the next user.

6.1.19 DTMF Suffix-Dialling for Attendant Consoles

This feature allows attendants temporarily switch over to DTMF signalling on open lines, in order to be able to access and use public network features or tone-controlled equipment via the public network, such as CityCall systems or other paging/code-calling systems, Voice Mail systems, etc. The digital signals sent by the attendant console are converted into DTMF tone signals by the line switching equipment (not by the attendant console). For outgoing calls, this can be controlled by attendants through dialling a "DTMF access code" in order to loop the connection through a signalling interface unit, or simply through selecting an appropriate route or trunk group already configured for DTMF conversion when setting up the call. In the case of incoming calls, DTMF conversion of attendant console signals can only be activated by dialling the DTMF code. The feature offers the full functionality of the DTMF suffix-dialling feature, as follows:

- The dialled digits (DTMF signals) and the route-dependent messages are output on the attendant console screen

- DTMF signalling can only be activated explicitly, i.e., on a per call basis
- The DTMF access code can be programmed on a repertory key
- Attendants cannot hear the DTMF tones
- The Pulse/Pause (make/break) ratio can be configured individually for each trunk group

The feature can be used by attendants in the following situations:

- during outgoing voice calls
  – automatically via line selection, i.e., without dialling an activation code
  – after "manually" dialling an activation code
- during incoming external voice calls
  – after "manually" dialling an activation code
- while setting up an outgoing voice call (while the call is still being rung)
- during consultation hold of an outgoing voice call or call setup (while the call is still being rung)
  – automatically via line selection, i.e., without dialling an activation code
  – after "manually" dialling an activation code
- during call extensions, either after the successful connection of the extended call with the destination user or while the extended call is still being rung through.
- after call transfer

6.1.20 Activating, Deactivating and Programming Call Forwarding at the Attendant Console

Attendants can carry out the following functions for a specific station user in the HiPath 4000 network via a code sequence procedure, from any attendant terminal (knowledge of the user’s PIN is not necessary):

- activate or deactivate variable call forwarding (FWD) to internal, external, or network destinations
- deactivate fixed call forwarding

This feature is not available for (standard) night service stations.

- The calling party is notified of the call forwarding, just as if the station user had activated the feature instead of the attendant. The option of a display suppression function for this feature is currently not implemented.

- The following call forwarding destinations are possible:
  - analog telephones
  - digital system-telephones
  - IP-telephones
  - hunting group
  - attendant's personal telephone no.
  - call queue
  - digital exchange (subscriber) line
  - coder calling system/paging system in meet-me mode

  The activation and deactivation procedures as entered by the attendant may produce different results, depending on the situation or status of the user's station.

  - Attendant enters deactivation code:
    - if station is not authorised for variable call forwarding, variable FWD or has not been activated: no change in telephone status.
    - if variable FWD is activated at station: variable FWD is deactivated.
    - if FWDBSY (call forwarding if busy) is activated: no change in telephone status, FWDBSY remains active.
    - if fixed FWD is activated at station: variable FWD is deactivated .
    - if fixed FWD destination is programmed for station, but call forwarding for all calls has not been activated: no change in telephone status.
  - Attendant enters activation code:
    - if variable FWD is not activated at station: variable FWD is activated.
    - if variable FWD is already activated at station: variable FWD destination is overwritten with new destination.
    - if FWDBSY is already activated at station: variable FWD of all calls is activated (FWDBSY destination remains unchanged).
    - if fixed call forwarding is already activated at station: variable FWD is activated to different destination (fixed FWD destination remains unchanged, and fixed FWD again becomes active when variable FWD is deactivated).
    - if fixed call forwarding destination is programmed at station, but fixed FWD is not activated: variable FWD is activated (fixed FWD destination remains unchanged; FWANS + FWDBSY becomes active when variable FWD is deactivated).

6.1.21 Call Tracing

- The tracing feature can be used to record the number of a caller on a printout in case of annoying calls (when calls do not identify themselves).
- The system must have the A call number available in order to trace calls (internal call, call from a satellite PABX, private network, ISDN).
- Automatic tracing of all calls (when ringing starts) during a certain period:
  - The administration and maintenance system is used to issue a tracing class-of-service.
  - The numbers of all callers are automatically printed for as long as tracing is activated at a station.
  - The tracing class-of-service can be issued to users and attendants.
  - The tracing class-of-service can be activated for a maximum of all station users/attendants at the same time.
- Manual call tracing
  - All users can cause the number of the caller to be printed out using a standard procedure.
  - Tracing is performed only for the call for which the procedure was selected.
  - Negative acknowledgment if the feature is not present or the number of station A is not available.
- Printout at a central printer (service terminal)
  - For tenant service, printout for all customers at a common printer.

6.1.22 Join (e.g. Extend call to paged user)

Attendant consoles are able to JOIN together two calls on different Loop keys (Lines) by activate the new JOIN key. This feature can be used for example to extend a waiting call on the Attendant console to a wanted paged user who calls back the Attendant :

- While a caller is held on the Attendant, the Attendant searches for the person called (= paged user) through the personal paging system.
- If the paged user calls the attendant console (using the personal AC call number), he/she can be connected to the held caller.
- Feature is not administrable. It is always available in the Attendant consoles.

6.1.23 Transfer a call to extension with activated call forwarding

This feature allows an attended to override a forwarding at the called destination.

6.2 Attendant functionality "Multiple Queing"

6.2.1 Multiple queues and parallel presentation of calls

The feature is administrable on HiPath 4000 Attendant Group level and decide the Attendant Group Mode for process incoming calls If the Attendant Group Mode is configured to Multiple Queues the Attendant of this Group must be the AC-Win MQ only.

Major characteristics

- Group Queues
- A total of twelve call queues are provided. The configuration specifies the queue into which a call is put (call type or network information dependant) and how it is placed there (call priority).
- All incoming voice calls (including personal calls) are queued to one of the twelve source queues.
- The following three parameters are defined for each type of call that the system can distinguish:
  - The number of the appropriate queue;
  - One of 8 possible priorities for the sequence within the queue;
- A queue time value for Priority Upgrade and color change of the call presentation can be defined for all calls on a system level.
- Charge calls to the Attendants or the Attendant Group are queued to an adiitional Charge call queue.
- Fifo behavior for every queue
- Time / volume controlled overflow to an administrable "overflow destination"
- Attendant Source keys
- For each queue and on each AC-Win MQ there is a source key for call presentation and call acceptance. Each attendant console has twelve source keys. The following information is presented:

- What type of call the key is used for: text label under the key according to the AC-Win MQ label configuration.
- The number of queued calls. There is no LED bar on if no call is queued.

- Call acceptance is performed by pressing the source key on any attendant console of the group.
- Attendant Loop keys
  On acceptance, the call is removed from the group queue and parallel source key signalization and assigned to the answering attendant console. Here it is looped to a special key (loop key) of the attendant console. The call on the loop key is then further handled at this attendant position and only shown on this display (not parallel on all attendant consoles).
- Each AC-Win MQ has six loop keys to process the calls. This means that six answered or established calls can be present on an attendant console at the same time.

6.2.2 Control of diversion

This feature makes it possible to call a subscriber through the attendant console even if the subscriber has activated call diversion. Call setup to a diverted subscriber is first held in an interim status. The operator receives display information for diversion and is prompted to either break through the diversion or to follow it. For this purpose the two key functions "RING" and "STEP" are available on ACWIN MQ. If RING is selected, then the diverted subscriber is called, if STEP is selected, then the diversion destination subscriber is called.

Key points

- Functions throughout the network (HiPath 4000)
- Works only for unconditional / immediate call diversion and not for call diversion no reply /busy
- Works only for the first call diversion (not for multiple call diversion)

6.2.3 Call retrival

This feature enables the attendant to retrieve extended calls which are not yet answerd back to the same attendant console. For this purpose there is a "Retrieve" key on the AC-Win MQ. After key pressing the attendant is again connected with the caller and can then set up the connection.

Generally the feature works:

- only for the last call transfered from this attendant console; destination subscriber free or busy,

- only if the transferred call to the wrong destination has not yet been answered or cleared.

The retrieved call is automatically switched to the next free loop key as soon as the "Retrieve" key is actuated and is immediately in call condition.

7 Networking (TDM- based)

Inter-Communication Server traffic is the handling of communication in communication networks in terms of call processing

- in different system configurations
- with different interfaces.

The system configurations are as follows:

- HiPath 4000 networking
- HiPath 4000 with other systems via tie lines.
- Expansions of existing systems with the addition of intermediate traffic.

The various interfaces are for example:

For the central office configuration

- analog DID incoming, not bothway
- digital DID incoming and outgoing
- ISDN interfaces

For the PABX-tie line configuration

- analog two or four-wire interfaces
- digital interfaces (channel-associated signalling)
- ISDN interfaces (S_0/S_2)

For the PABX-main/satellite PABX configuration

- analog two-wire interface
- digital interfaces with chanel oriented signalling
- ISDN interface (S_0/S_2)

7.1 Networking

The term networking covers the functionality of main system connections, tie line configurations and digital CO trunks.

The systems in a HiPath network can be administered and maintained from HiPath 4000 Manager. The functions of the HiPath 4000 Managersare presented in a separate description.

Many features that operate throughout the network can be set up or barred individually for each line/trunk group:

- Attendant intercept in the event of
  - incomplete dialling.
  - user not available,
  - user busy,
  - not classmarked,
  - all trunks busy
  - do-not-disturb,
  - timeout
- Override
- Call waiting
- Transfer (in the event of busy, ringing, or call state, or only in the call state)
- Call forwarding - all calls
- Call forwarding - no answer
- Callback
  - automatic
  - automatic - no answer
- Suppression of recall to the ATND in nodes without an ATND

Please refer to the AMO descriptions in the service manual for detailed information on features that can be influenced.

7.1.1 Digital CO Trunks

Connection via basic access S_0 or primary rate access S_2M
(D channel protocol to ITU: Q.930/931

1. Features
   - DID for outgoing and incoming trunk calls
   - Outgoing trunk calls extended by attendant console or night station
   - Incoming trunk calls extended by attendant console or night station
   - Consultation hold from outgoing or incoming trunk calls; only via 2nd line throughout network

- Transfer or pickup of outgoing or incoming trunk calls
- Signalling of trunk calls
- Call data registration for outgoing trunk calls
- Display of the calling party's call number at the called party (attendant console/digital voice terminal)
- Suppression of call number display (unlisted call number)
- Service signals
- Use of special facilities
  - paging system
  - recorded announcement equipment
- Permanent activation of layers 1 and 2 for S_0/S_2
- Service indicator
- Simultaneous access
- Call forwarding (all calls) to users in the public exchange and to users at systems behind the public network, subject to the following restrictions:
  - Voice connections only
  - Only with special classmark
  - Connection from ISDN exchange via HiPath 4000 to ISDN exchange is possible, prevention can be set by means of a class-of-service
  - Charges for connections from the system to the public exchange borne by the forwarder of the call
  - Max. no. of external digits: 22
  - User interface at the systems terminal identical to that for a call forwarded internally within the network
  - Add-on conference from consultation hold via 2nd line

Main/Satellite PABX Calls

1. Main PABXs
   - Incoming/outgoing connection to the central office (public network)
   - Attendant console functions for main and satellite systems
   - Call data registration for all satellite system users
2. Satellite systems
   - No incoming trunks

- No attendant console
- The number of satellite systems is limited only by the number of free mounting locations in the main or satellite system and the directions that can be set.
- No attendant console service for outgoing lines,

3. Basic features between main and satellite systems
   - Connection setup for users
   - DID outgoing to satellite system users
   - DID incoming for satellite system users
   - Connection setup to the attendant console of the main system for satellite system users
4. ISDN interface (S_0/S_2)
   - Connection via basic access S_0 or primary multiplex connection S_2 with the CorNet NQ signalling protocol
   - The transmission equipment must be ordered and installed separately.

Features for S0/S2 CorNet NQ signalling

- Internal traffic incoming and outgoing from/to the remote system
- Consultation hold from outgoing/incoming internal calls
- Transfer of an outgoing/incoming internal call through call pickup or call transfer.
- Transfer of a trunk call from main stations through call pickup or call transfer.
- Prevention of call transfer if transfer of calls has been prevented via AMO in one of the PABXs involved in the connection.
- Special consultation hold (alternating between parties)
- Use of toll/code restriction f
- Use of the central RNG
- Special ringing for calls
- Call data registration
- Call data registration at the attendant console with display of the station number at the attendant console
- Busy override/camp-on
- Call number (and name) display

- Suppression of number (and name) display for standard and confidential station (suppression of display)
- Use of special private facilities
  - dictation equipment
  - public address systems
  - recorded announcement equipment
  - Handsfree talking facility
  - paging systems
- Automatic callback
- Automatic callback - no answer (also from mailboxes)
- Call forwarding - all calls/call forwarding - no answer to user, server and paging system with multiple paging (meet me) in foreign nodes (network-wide)
- Add-on conference from consultation hold via 2nd line
- Add-on witness
- Outgoing connection setup in the network via
  - dial circuit and terminal dialling at the service-specific terminal
  - hot-line (direct station selection)
- Incoming DID call for subscriber from the network
- Service change via S_0 bus interfaces
- Multi-service operation
- Hunting group
  - Network-wide availability, but no configuration of a hunting group with subscribers in different networks.
- Central attendant console

7.1.2 Tie Traffic

(Traffic between similarly authorised systems in the network)

7.1.2.1 Digital interfaces

Digital tie traffic is handled with DIUC (Digital Interface Unit with Channel Associated Signalling) in accordance with PCM30 recommendations (e.g. G.703).

- Interface HDB3, 120 Ohm or 75 Ohm
- Scope of signalling as for TMEMW
- Configuration as master/slave with system clock synchronisation is possible

7.1.2.2 ISDN interfaces

ISDN S_0/S_2 interfaces complying with the CorNet-NQ protocol are available with the same functionality as described for the ISDN interface for main/satellite PABX traffic.

7.1.3 Non-Voice Features

The following network-wide non-voice functions are available irrespective of the configuration (tie traffic, inter-Communication Server traffic).

For transit traffic a transition from S_0/S_2 to a/b is program. In this case, the features valid for the a/b lines shall apply.

1. Principal non-voice features available throughout the network:
   - Valid for all terminal types via S_0/S_2 networking interfaces:
     – outgoing connection setup
     – incoming connection setup (DID)
     – multiple trunk group with overflow
     – routing trunk groups with preset digits
     – classes-of-service for inter-Communication Server traffic per service
     – speed calling
     – hunting group (network-wide availability, but no configuration of a hunting group with subscribers in different networks)
     – call forwarding
     – camp-on and busy override protection
     – closed numbering throughout the network
     – prevention of inadmissible or undesired calls

   – network-wide signalling of the call partners' numbers
   – suppression of display
   – service change via S_0 bus interfaces
   – multi-service operation

2. Additional features for the following terminals and interfaces:

   Data service:

   DTE with modem (networking interface S_0/S_2):

   Connection destinations:

   - in private network
   - in analog telephone exchange
   - in ISDN
   - digital DTE internal and external

   Facsimile service:

   FAX Group 2/3 terminal (networking interface S_0/S_2 and a/b):

   Connection destinations:

   - Fax terminal in private network
   - Fax terminal in ISDN
   - Fax terminal in analog telephone exchange

   Facsimile Group 4 terminals (networking interface S_0/S_2):

   - Facsimile terminal in the private network
   - Facsimile terminal in ISDN

Non-voice-connections via DPNSS 1/DASS2-CorNet Gateway

HiPath 4000 offers the possibility of interconnecting digital networks, provided the CorNet NQ and DPNSS1 or DASS2 protocols are used for transmission. The possibility of transmission is provided with the aid of "DPNSS1/DASS2-CorNet-Gateway" (CDG). The CDG translates CorNet NQ messages into DPNSS1/ DASS2 messages and vice versa.

The following digital network combinations are possible:

DASS2 as transit:	HiPath 4000		DASS2		HiPath 4000
DPNSS1 as transit:	HiPath 4000		DPNSS1		HiPath 4000
HiPath as transit for DPNSS1:	DPNSS1		HiPath 4000		DPNSS1
HiPath between DPNSS1 and DASS2:	DPNSS1		HiPath 4000		DASS2

HiPath after DASS2:	HiPath 4000		DASS2		
HiPath after DPNSS1:	HiPath 4000		DPNSS1		

7.1.4 Special Facilities

Network-wide application possible via the a/b, S_0/S_2 interfaces. Special facilities are connected via the paging system (TMOM). The following devices and their service features are available throughout the network.

1. Recorded announcement equipment
   - General announcement and announcement with greeting text
2. Public address system
3. Entrance telephone
   - door conversation
4. Paging system
   - Simple paging system
     – with voice announcement with/without talk-back facility
     – with display
     – with answering callback (meet-me):
     network-wide paging,
     network-wide answering, except via CO trunks.

   The dialling aid keys be used locally and on a network-wide basis to activate this feature. The following paging variants are possible:

     - Callup of a complete digit series
     - Callup of a partial digit series with subsequent manual dialling of the remaining digits
     - After initial manual dialling, callup of the remaining digits using a dialling aid key.

   Simple paging can also be set up as a hotline or off-hook recall destination.

   - Multiple paging
     – with display
     Users connected to a multiple paging system can also be paged in consultation if the following conditions are met:
     – 'display' operating mode,
     paging/paged user in one node.

This feature can also be implemented on a network-wide basis if the code-calling party and the CCM are in the same node (A), the called (paged) party is in the node (B), the called (paged) party replies from a telephone in the node (B) and the nodes (A) and (B) are linked via S0/S2 lines

The answering procedure is only possible from an originating call. The feature does not apply to the attendant console.

– answering callback (meet-me):

– paging network-wide,

– answering network-wide, except via CO trunks.

The display does not go to lines which do not supply a calling party call number (analog exchange). It applies only in certain cases to ISDN exchanges because the total number of digits supplied (comprising calling party call number + block code + called party call number + block code) may exceed 22 digits. The paging equipment must be able to handle variable A block.

The "meet me" function works network-wide only if the paging user lifts the handset and waits.

7.1.5 Specific Displayinformation

7.1.5.1 Displays on digital system- / IP-telephones

- For calls which are forwarded in the remote system (FWD, CHESE, HG) or transferred, the calling party will see on his display the call number he has dialled instead of the name of the user.
- If a caller is forwarded by the forward-on-no-answer feature or within a hunting group, the caller's display is not updated while the call is being forwarded. This means that the caller will not realise that his or her call is being forwarded until it is answered, when the name and/or number of the call forwarding destination is output.

7.1.5.2 Displays on the attendant console

- For trunk calls and attendant calls the record type is sometimes displayed instead of the name of the user dialled in the remote system:
  – User busy
  – User with do-not-disturb feature

  - User not classmarked
  - User not obtainable

- For calls which are forwarded in the remote system or transferred, neither the name of the dialled user nor the dialled call number is displayed. In contrast to terminal traffic, an existing FWD is displayed for outgoing trunk calls.
- In the case of transfer of ringing and transfer of ringing in a hunting group, the display at the attendant console is not updated until the user in the remote system answers.

7.1.6 Central Attendant Console (CAC) in a System Network

7.1.6.1 Definition

- The "normal" ATND is also used as central ATND (CAC).
- An ATND is always assigned to one ATND group (ATNDGR) exclusively.
- An ATNDGR is an answering unit for "console traffic". Normally, an ATNDGR includes one or several ATNDs for day-time operation and up to 8 night switching variants of various night switching types. The characteristic feature of an ATNDGR, however, is the separate general attendant.
- A system can have several ATNDGRs.
- Each ATNDGR can be reached unambiguously in incoming console traffic. Selection takes place by means of the console or answering code or indirectly via the internal traffic restriction or digit analysis group allocation.
- If an ATNDGR also handles the console traffic of other ATNDGRs (of other systems of a network or of own system) temporarily or permanently in addition to own (local) console traffic by means of transfer, it operates as "central ATNDGR" (CACGR) for this ATNDGR by definition.
- Every ATND of a CACGR is a central ATND (CAC). It does not differ from a "normal" ATND as far as equipment and features are concerned.
- Several CACGRs can be installed and activated simultaneously both in a system network and in a system.
- A CACGR can also be operated in night switching mode without an ATND being installed ("central night station").

7.1.6.2 Global functions

- Systems in the network which have neither an ATND nor a night switching facility are serviced by an ATNDGR of another system as far as "console traffic" is concerned (= CACGR for the serviced system).
- ATNDGRs with ATND can temporarily change over the console traffic to an own night switching variant or alternatively to a CACGR of another system in the network or own system. The selection is to be performed by means of a procedure on the ATND.
- The possible allocations of an ATNDGR to CACGRs are to be created via the service terminal (analogously to create and select a night variant).
- Changeover can also be performed automatically if the last ATND of an ATNDGR does not answer a call (analogously to automatic night switching).
- Joint queues in a CACGR for the local calls and the calls of all the ATNDGRs serviced by it (even waiting times).
- A CACGR can change over, in turn, to a system-internal night switching variant.
- The ATND or night extensions of a CACGR must be able to reach all subscribers and ATNDs of the system network and extend calls to them.

7.1.6.3 Incoming calls

The following types of call are possible:

- First calls via various console code numbers and digit analysis group (direct inward dialling and main station interface)
- Attendant intercept
- Recall
- Serial call
- Consultation request
- Absent XFER calls

Signalling at the call keys

- Trunk calls for the A-call key
- Other calls for the M-call key

Remark: Only those calls received from the call queuer are forwarded to the central attendant console, in other words not any calls for the P-call key of an attendant console. At the P-call key, each central attendant console is only accessible network-wide via its own call number.

Recurring calls

Calls from extended connections arrive over and over again at the attendant console which extended them.

Signalling in the call line

- Type of call (FIRST CALL etc.)
- Type of connection (TRUNK etc.)
- Name (if caller in the network or from outside via digital line), otherwise: trunk group designation
- Company name (via console code or via subscriber or trunk ITR group in the case of call intercept)

Signalling in the answer display

- Source:
  – Call number (or trunk group designation)
  – Name (if any)
- Destination (in the event of recall and call intercept):
  – Destination number and name
  – Reason for intercept or recall

Taking account of the priority in the call queuer

- According to the type of call and connection
- The priority-level allocations fixed for the central attendant console group apply (not those for the source attendant console group)

7.1.6.4 Extension of incoming calls

- To all users in the integrated network
- To all attendant consoles in the integrated network (transfer to P-call key)
- To users in the free and busy states
- Busy override
- Verification (also for internal calls)
- Taking network-wide account of the ITR group switching security
- To users with user features within the system (call forwarding - all calls, call forwarding - no answer, hunting group, call pickup group, exec./secretary telephone)

- To users with global features (call forwarding - all calls, call forwarding - no answer)

Signalling in the answer display

- Dialled number
- Name of dialled subscriber
- State of dialled destination
- Name and number of forwarding destination:
  – immediate, for call forwarding - all calls
  – after answering, for call pickup group and exec./secr.
  – after forwarding, for call forwarding on no answer and hunting group

Setting up outgoing calls

- The attendant console can obtain every user in the network
- The attendant console can extend or assign external connections to every station of the network provided the "Least Cost Routing" feature is implemented in each node of the network.

Signalling in the answer display (source)

- Dialled number
- Name of dialled subscriber
- State of dialled destination
- If appl., name and number of forwarding destination:
  – immediate, for call forwarding - all calls
  – after answering, for call pickup group and exec./secr.
  – after forwarding, for call forwarding on no answer and hunting group

Connection paths

- Extending: The system to which the user belongs is not taken into account when searching for the circuit
- Assignment: The system to which the user belongs is taken into account, i.e. preference is given to seizing a circuit (e.g. a trunk circuit) in the system to which the user is connected.
- It is not possible to book a line or reserve lines on an inter-system basis.

Switching

- As for incoming traffic

- it must be possible to prevent exchange - exchange connection
- no speed servicing or extending of calls with announcement during multiple paging (meet-me) before the paged user answers.

7.1.6.5 Check-key functions

The following features can only be activated and deactivated by user groups in the home system:

- Do-not-disturb
- Station group restriction
- Class-of-service switchover

As previously in individual systems, the remaining check key functions are limited to the ATNDGR or affected attendant consoles.

7.1.6.6 Alarms

Only the alarms for the home system are indicated at the attendant console.

7.1.6.7 Activating and deactivating a CACGR

Transferring the console traffic from an ATNDGR to a CACGR is to be considered as a new night switching type.Like the ATNDGR night variants, this transfer can also be created by means of administration and maintenance and finally assigned to one of the variants 1-8 of an ATNDGR.

One of the up to 8 variants is selected, i.e. "predetermined", on the ATND by means of a check procedure using the cursor. This "predetermined" variant becomes the "current" variant, i.e. effective, when the last ATND of the ATNDGR is taken out of operation (e.g. by pulling the jack).

A variant can also be "predetermined" on the service terminal (this is the only possibility for predetermination in ATNDGR without ATND).

Transfer to the CACGR is also possible in connection with "automatic night switching". For this purpose, the forwarding destination must be entered in variant 8.

In the event of changeover to a CACGR, the calls waiting in the general attendant at this point of time will also be transferred to the CACGR.

A CACGR must not redirect calls to another CACGR in another node.

Within a node it is possible to transfer a CACGR's calls to another CACGR on a single or multi-stage basis by means of automatic changeover. In the case of multi-stage transfer, there is prevention of loop formation in the node. The original configuration is automatically re-established on cancellation of changeover.

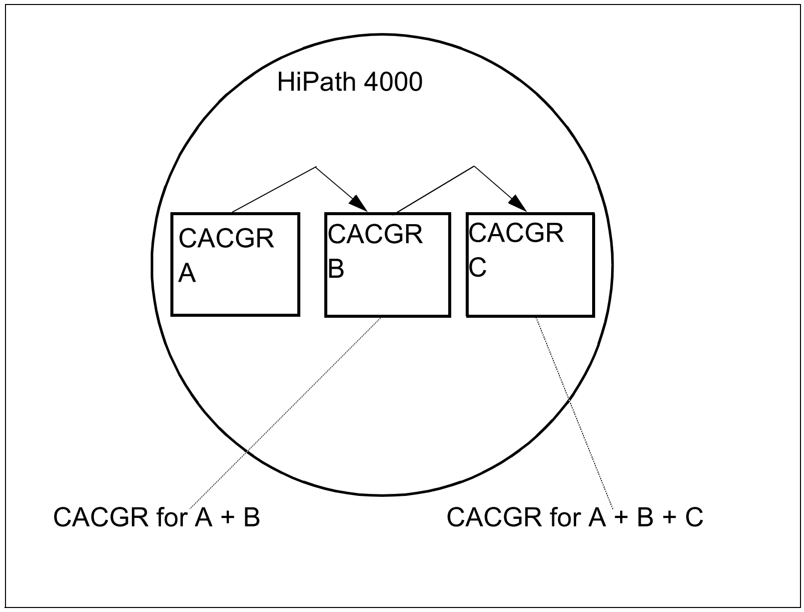


Figure 27 Principle of transfer with 3 CACGRs

Forwarding calls to a central attendant console group

Every call destined for the attendant of the system (from a user or external circuit) is through-connected to the home call queuer (general attendant), including calls to systems without an attendant console or night service.

Depending on the "current option" (in the dynamic device memory of the call queuer), either an attendant console or a night option of the relevant attendant console group is seized, or alternatively the call is transferred to a central attendant console group.

In the latter case, the general attendant sends the required data (direction, ATND code) back to the caller (trunk/subscriber) who, in turn, transmits the call to the respective CACGR.
If the call can be transmitted in the remote system, the caller receives a positive acknowledgment.

In the event of a negative acknowledgment (e.g. blocking the data path to the other system), the seizure attempt is repeated at intervals of a few seconds until the attempt is successful.

Changeover to a central attendant console group

If an attendant console group is placed out of service (e.g. by disconnecting the jack at the last attendant console), the set option – which is option 8 for automatic night service switching – becomes the "current option".

Cancellation of forwarding to a central attendant console group

Forwarding to a central attendant console group is cancelled if the first attendant console in the forwarding attendant console group either inserts the jack (changeover to day option) or changes over to another night option.

Variant definition

1. Variant types (alternative to day-time switching)
   - night switching in own system
     - night ATND
     - night extensions
     - universal answering
   - empty variant
   - Transfer to a CACGR
     Note:
     - Variant type c) is not possible for a CACGR.
     - Universal answering is limited to subscribers of own system.
2. A maximum of 8 variants of these types are created by administration and maintenance for each ATNDGR (numbers between 1 and 8);
3. the variant marked (predetermined) with the cursor on an ATND of the ATNDGR becomes the current variant after pulling the jack on the last ATND of an ATNDGR.

Administration and maintenance

The possible forwarding destinations (system and ATNDGR) for each system are to be created on the service terminal. The same applies to the assignment of these forwarding destinations to the individual variants of an ATNDGR.

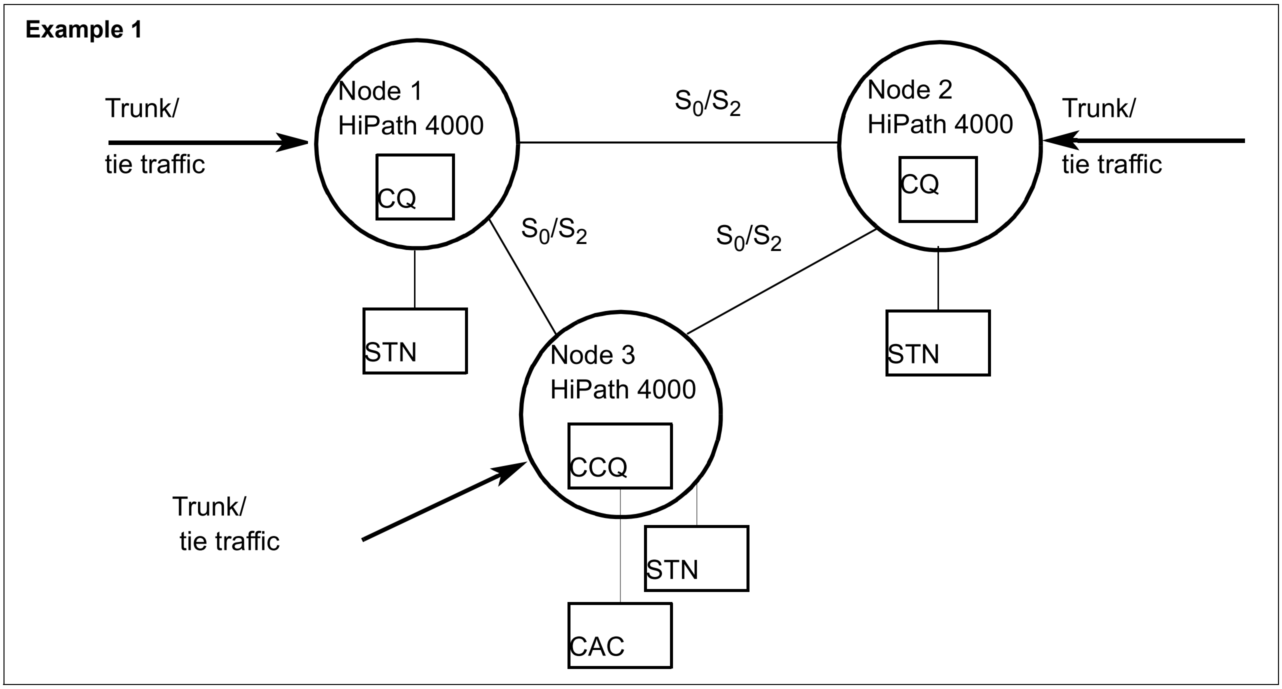
Dependability

If the path or the system of the CACGR is faulted (no positive acknowledgment), the call is transferred to ATNDGR no. 16 of own system after several unsuccessful seizure attempts. ATNDGR no. 16 may be set, for example, to a system-internal night extension or to an alternative CACGR in another system.

7.1.6.8 Distribution of incoming calls in system teaming

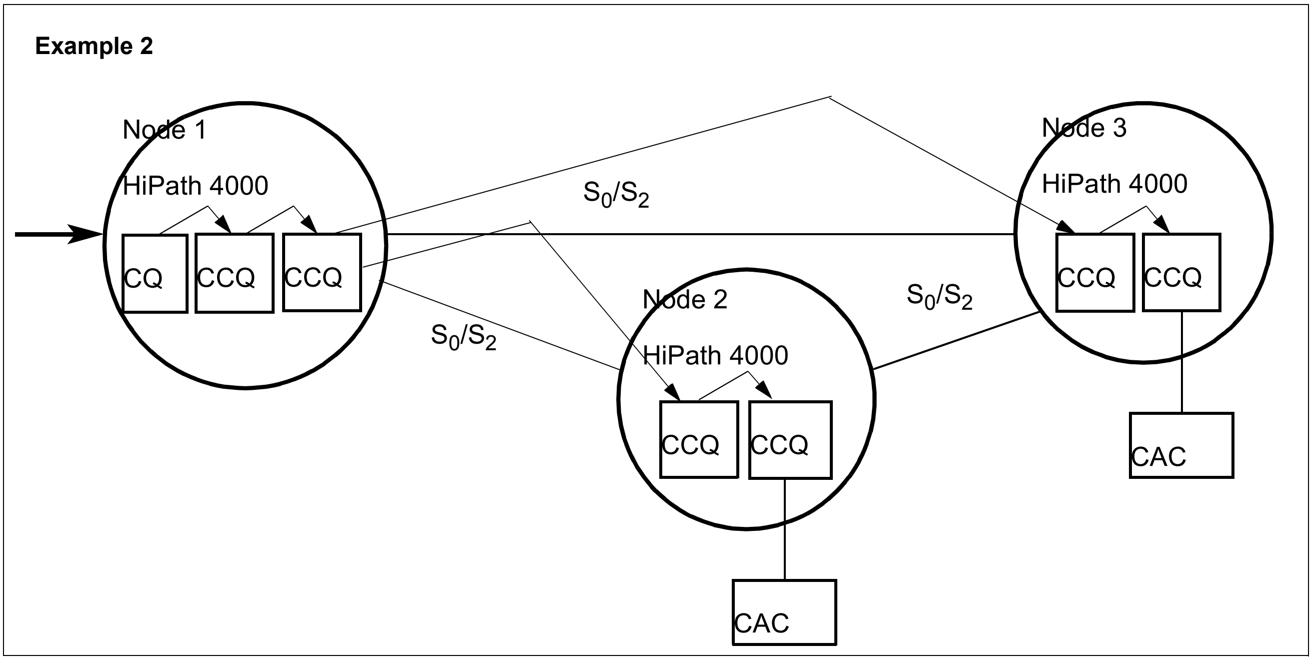
The CAC (central attendant console) is intended for use in a digital S_0/S_2 network. The CAC functions can only be used in a homogeneous HiPath 4000 network .

- each HiPath 4000 node has at least 1 up to a maximum of 16 call queuers (CQ)
- calls to the attendant each go to the AOs in the node in which they originated
  - calls from internal users (station calls to attendant)
  - external calls via digital/analog CO trunks and tie lines at the node



Explanations:

- Traffic to the attendant from node 1 and 2 is transferred by the relevant CQ to the CCQ or external CAC in node 3.
- Transfer of traffic to the attendant from node 1 takes place directly via the S_0/S_2 link to node 3 or (in the event of overflow or if no connection exists between node 1 and node 3) via node 2 (transit); the CQ of node 2 is not occupied in the case of this transit traffic.
- The same applies analogously to the traffic to the attendant from node 2 to node 3.



Explanations:

– Traffic to the attendant from node 1 can be transferred to either node 2 or node 3, depending on the NAVAR setting in CCQ 2.

– It cannot be transferred from node 2 to node 3.

– It can be flexibly transferred within the nodes.

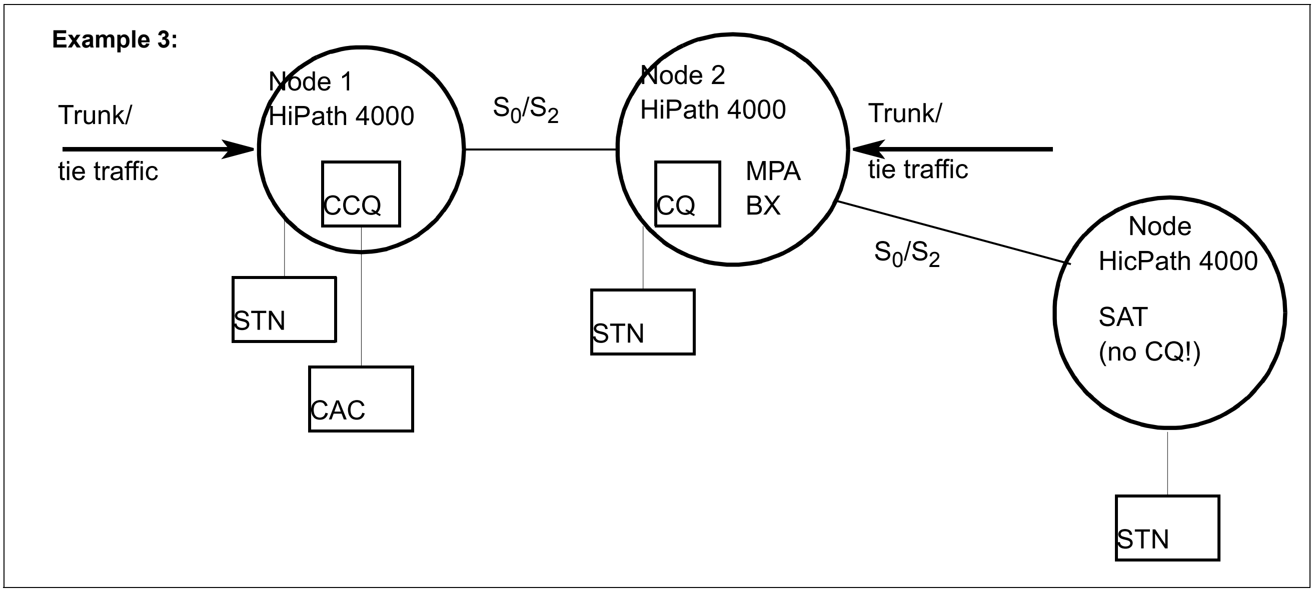
– Transfer from node 1 to node 3 can take place via a direct route or via node 2 (transit); The CCQ (central call queue) in node 2 is not occupied in this case

- The CQ can transfer calls to a central call queuer (CCQ) of any node (external CAC) or its own node (internal CAC), depending on which night variant has been set (NAVAR) (see example 1)
- However, a call only be transferred once from node to node; within a node, transfer from CCQ to CCQ can take place as often as required (see example 2).

Special cases: HiPath 4000 node as a satellite system in the homogeneous digital network

It is possible to operate a node without a separate CQ, namely as a satellite system (SAT) connected to a main system (MPABX) (see example 3).

Features of the satellite system: no external trunk, no attendant console



Explanations:

– Calls to the attendant from the satellite system go to the CQ of the main system.

– In the example shown, calls to the attendant from the CQ of the main system (node 2) are transferred to the CAC in node 1; calls to the attendant from the satellite system (service calls, call transfer to attendant) can thus de facto be routed twice from node to node because the CQ in node 2 is in this case not a CCQ.

7.1.7 Preventing Network Loops

- Each node of a networked system must be assigned an unambiguous node access code.
- In networked systems with CorNet NQ signalling, the access code of the source node is transmitted with the calling line identification message in each call setup request, and checked against the destination node access code in each transit node via the number matrix.
- As a preventive measure against trunk group blocking caused by network loops, and in addition to the verification of the source node access code against the destination node access code, a hop counter is maintained.
  – A hop counter is maintained for each connection setup
  – When the connection is set up, each transit node ("hop") is counted and the cumulative total is reported to the next node in the call setup message.

- As soon as the threshold value is reached, the connection setup is aborted.
- The hop counter overflow is displayed on the service terminal.
- The hop counter is signalled via the CorNet-NQ and DPNSS1 protocols.
- For DPNSS1 signalling, the hop counter is converted in the CDG.
- The hop counter threshold value can be configured separately for each trunk group via AMO.

- In order to limit the number of transit nodes used in a connection, a transit node counter is also maintained. Like the hop counter, the transit node counter can terminate the connection setup when a defined threshold is reached, and the overflow is displayed on the service terminal.

7.1.8 Heterogenous networking using QSIG, PSS1

QSIG = "Signalling protocol at the Q reference point" (designation in the ETSI standards) or PSS1 = "Private Signalling System No.1" (designation in the ISO/IEC standards) are the internationally standardized "Inter-Exchange Signalling Protocol", i.e the internal networking protocol for corporate networks.

The product QSIG is released for HiPath 4000. At the same time as QSIG was being implemented, the proprietary Siemens networking protocol CorNet-N was enhanced to become the standard-based networking protocol CorNet NQ:

- CorNet NQ is a real superset of QSIG/PSS1, in as far as it is implemented in HiPath systems.
- Existing CorNet NQ features not standardized for QSIG/PSS1 or not yet implemented according to the standards, are proprieatary features and as such "tunneled" through QSIG/PSS1, using the standardized QSIG/PSS1 container mechanism.
- New networking features not standardized for QSIG/PSS1 are defined within Siemens according to the QSIG/PSS1 specification rules.

7.1.8.1 Brief service description

HiPath 4000 supports the connection setup via QSIG/PSS1 for all Circuit-mode Bearer Services and Circuit-mode Teleservices used in public Euro-ISDN.

The following Circuit-mode Bearer Services are supported:

- 64 kbps Unrestricted Bearer Service
  64 kbit/s Bearer Service

- 64 kbps Service usuable for 3.1 kHz Audio Information Transfer
  3,1 kHz Bearer Service
- 64 kbps Service usuable for Speech Information Transfer
  Speech Transfer Service

HiPath transfers all other Circuit-mode Bearer Services in transparent mode.

The following Circuit-mode Teleservices are supported:

- Telephony 3.1 kHz
- Telephony 7 kHz
- Telefax Group 4

HiPath transfers all other Circuit-mode Teleservices in transparent mode. Fallback procedures for the teleservices Telephony 7 kHz and Videotelephony are not supported.

7.1.8.2 Brief feature description

In the QSIG/PSS1 standards the features are divided into Supplementary Services (SS) and Additional Network Features (ANF):

- – Supplementary Services can be used directly.
- – Additional Network Features are additional features used within the network. The user cannot use or influence them directly.

1. **Number Identification** Supplementary Services

   Calling Line Identification Presentation (CLIP)

   CLIP allows the signalling and display of the calling party‘s number at the called party.

   Connected Line Identification Presentation (COLP) (Transmission of B party’s number to A party)

   COLP allows the calling station to have the number of the called party displayed.

   Calling/Connected Line Identification Restriction (CLIR) (Suppress transmission of A party’s number to B party and vice versa)

   CLIR allows

   - – CLIR allows a calling party to suppress the display of his number and, if used, a subaddress (SUB) at the called party.
   - – a called party to suppres the display of his number and, if used, his/her subaddress (SUB) at the calling party.

Suppression can be activated either in permanent or in temporary mode, i.e. on request of the calling party.

2. **Name Identification** Supplementary Services

   Calling Name Identification Presentation (CNIP) (Transmission of A party‘s name to the B party)

   CNIP allows signalling and display of the calling party‘s name at the called party.

   Connected Line Identification Presentation (CONP) (Transmission of B party´s name to the A party)

   CONP allows the calling station to have the name of the called party displayed.

   Calling/Connected Name Identification Restriction (CNIR) (Suppress transmission of A party’s name to B party and vice versa)

   CNIR allows

   - a calling party to suppress the display of his name at the called party.
   - a called party to suppres the display of his name at the calling party.
   Suppression can be activated either in permanent or in temporary mode, i.e. on request of the calling party.

3. **Call Completion** Supplementary Services

   Call Completion to Busy Subscriber (CCBS) (Callback on busy)

   CCBS allows a calling party to activate automatic callback if the called station is busy. As soon as the called party goes on-hook, the station initiating callback is called. When this station answers, a new call to the original destination is started.

   Call Completion on No Reply (CCNR) (Callback on no answer)

   CCNR allows a calling party to activate automatic callback if the called station does not answer. As soon as the called party establishes and finishes a new call, the station initiating callback is called. When this station answers a new call to the original destination is started.

   The following callback options are implemented in HiPath 4000 :

   – Service Retention Option (defines, whether the callback request remains active or is implicitly deleted, if the B party is again busy)

   – Path Reservation Option (defines, whether the path for callback is established before the A party is called or after the A party answers)

   – Connection Retention Option (defines, whether or not a signalling connection remains active during callback monitoring)

   – Idle Guard Timer (is started when the B party becomes idle. The B party is signalled idle to the A party after the timer expires.)

– This timer is not implemented in HiPath 4000. The B party is signalled idle immediately.

4. **Call Diversion** Supplementary Services

   Call Forwarding Unconditional (CFU) (Permanent/immediate call forwarding)

   CFU is used for call forwarding of all incoming calls to a station number defined by the user.

   Call Forwarding Busy (CFB) (Call forwarding on busy)

   CFB is used for forwarding of incoming calls to a station number defined by the user, if the originally called station number is busy.

   Call Forwarding No Reply (CFNR) (Call forwarding on free/on no answer)

   CFNR is used for forwarding incoming calls to a station number defined by the user, if the call is not answered within a configured period of time.

- **Call Deflection** (CD) (Call forwarding during ringing)

  A station can forward incoming calls to a different station during ringing by activating CD.

  Special conditions:

  – A ’Diversion Counter’ is configured for QSIG, counting the number of times a call was forwarded. If this counter exceeds a configured limit value, the call forwarding is suppressed and the original station is called.

  – The existing COT parameter (FNAN) is used to ascertain, whether this service is released on the trunk (i.e. if this service is implemented on the external PABX).

  – A new COT parameter (CFVA) was introduced to ascertain, whether the forwarding destination can be validated.

5. **Call Transfer** Supplementary Service (CT) (Transfer of a connection)

   A user with a call on hold and an active call, can connect the two parties by call transfer.

   Special conditions, restrictions:

   – Only "Call Transfer by Join" is offered at the Q reference point. "Call Transfer by Rerouting" is not implemented.

   – If Call Transfer is used with QSIG, then QSIG only supports "Transfer into Active", "Transfer into Ringing" and "Transfer into Busy", but not "Transfer into Dialing".

   – In HiPath 4000 networks, one of the involved end nodes initiates a Path Replacement after CT, as soon as the resulting call has reach the "Active" state. The Path Replacement is used for Route Optimisation. The new voice path is established from the calling party to the destination.

– In HiPath 4000 PABX Call Transfer is closely related to Call Hold. One of the two calls involved in the transfer is always in Hold. The station which was on hold before the transfer will become the "Paying Party" after the transfer. With QSIG, none of the two calls involved in the transfer must be on Hold. However, HiPath 4000 expects the orderly termination of a hold condition (i.e. by signalling of the appropriate message).

6. **Call Offer** Supplementary Service (CO) (Active camp-on)

   CO allows a calling party to initiate accoustic or optical camp-on, if the called station is busy. The called station can accept the waiting call, reject it (set the call to busy) or ignore it.

7. **Advice of Charge** Supplementary Services

   Advice Of Charge during the Call (AOC-D) (Transmission of call charges during a call)

   If AOC-D is active, the accumulated call charges are transmitted to the station in defined intervals either as call charge units or amounts with currency designation. Both call charges received from the public ISDN and and call charges generated by the HiPath 4000 are transmitted.

   Advice Of Charge at the End of the Call (AOC-E) (Transmission of call charges at the end of a call)

   If AOC-E is active, the total amount of accumulated call charges (call charge units or amounts with currency designation) is displayed at the end of the call. Both call charges received from the public ISDN and and call charges generated by the HiPath are transmitted.

   Restrictions:

   – "Free running" call charges are not supported, i.e. AOC-E messages without any assigned B channel connection are not taken into account.

   – AOC-S is not supported.

8. **Path Replacement** Additional Network Feature (PR) (Route optimization)

   PR is used for route optimization e.g. after Call Transfer. This avoids unnecessarily seized B channels and network loops.

   After Call Transfer one of the involved end nodes can initiate a Path Replacement, as soon as the resulting call has reached the ’Active’ state. The new voice path is established from the calling party to the destination.

   Restrictions:

   – If Path Replacement is requested for one path, the entire path must be replaced. It is not possible to maintain parts of a path. If no path to the destination is available, the original connection remains and PR is not executed.

   – Currently, Path Replacement only affects Call Transfer.

9. Addressing / Private Numbering Plan (PNP)

   Stations in different networks with different numbering plans, which are supposed to reach each other, can do so by conversion of their station numbers. For this purpose, HiPath 4000 supports private numbering plans as per ETSI and ISO standards.

   The support of private numbering plans allows interoperability for all Supplementary Services.Interoperability is possible not just between proprietary heterogenous PABX within a private network, but also between private networks with different structures as well as public networks. Harmonization of the numbering plans in the various networks is not necessary.

   Special conditions:

   – For numbering plans in private networks the HiPath 4000 uses a structure containing domain, subdomain and node numbers.
   – All received numbers are converted to implicit numbers for internal processing.
   – All numbers to be transmitted are converted to explicit numbers.

7.1.8.3 Certifications

To maintain the reliability and high quality in HiPath 4000 networks, external Communication Servers must be certified before they can be connected to the HiPath QSIG/PSS1 interface. The certifications have to be done by a Siemens certification test laboratory. Normally, protocol tests in the laboratory and field trials are required.

Already certified foreign products must be re-certified in the following cases:

- upgrading of a HiPath 4000 to a new software version.
- if a new version of the foreing product is used.

7.1.9 CorNet NQ - DPNSS1 Interworking with CDG

IMPORTANT: Note: The CDG ist not released in Germany.

The CDG (CorNet NQ - DPNSS1 Gateway) provides CorNet NQ interworking with DPNSS1 or DASS2.

Through the CDG line card, interworking between a number of features of DPNSS1 and CorNet NQ is guaranteed

Sections of BTNR 188	HiPath 4000 Support	Comment
0. Forward & Contents	Xes	
1. General	yes	
2. Physical Layer	yes	
3. Data-Link Layer (Level 2)	yes	
4. Message Types & Formats	yes	
5. Level 3 Signalling Procedures	yes	
6. Simple Telephony	yes	
7. Circuit-Switched Data Call	yes	
8. Swap	Transit node only	
9. Call Back When Free	yes	
10. Executive Intrusion	yes	
11.1 Call Forwarding Unconditional	yes	
11.2 Call Forwarding Busy	Partial	
11.3 Call Forwarding No Reply	yes	
12. Hold	yes	
13. Three Party	yes	
14. Call Offer	yes	
15. Non-Specified Information	yes	
16. Service Independant Strings	Partial	
17. Call Waiting	yes	
18. Bearer Service Selection	Transit node	
19. Route Optimisation	Partial	This is only supported on HiPath 4000 at transfer, not on alternative routing and for voice service only.
20. Extension Status	Transit node only	
21. Controlled Diversion (for Operators)	yes	
22. Redirection	yes	
23. Series Call	transit node only	
24. Three-Party Takeover	transit node only	
25. Night Service (for Operators)	yes	
26. Centralised Operator	yes	
27. Traffic Channel Maintenance	transit node only	
28. Remote Alarm Reporting	transit node only	

Sections of BTNR 188	HiPath 4000 Support	Comment
29. Add-On Conference	transit node only	
30. Time Synchronisation	transit node only	
31. Call Back When Next Used	yes	
32. Do Not Disturb	partial	
33. Remote Reg. of Diversion	yes	
34. Remote Reg. Of Do Not Disturb	transit node only	
35. Priority Breakdown	transit node only	
36. Call-back Messaging	transit node only	
37. Loop Avoidance	yes	
38. Forced Release	transit node only	
39. Text Message	transit node only	
40. Charge Reporting	transit node only	
41. Network-Address Extension	transit node only	
42. Call Park	partial	
43. Call Distribution	transit node only	
46. Call Pick-Up	yes	Directed Call Pickup - HiPath CorNet NQ feature , DPNSS InterworkingI

7.1.9.1 Mixed network enhancements

The following features concerning the interworking via CDG are realized within the HiPath 4000 system.

– Configurable Routing of Realitis- DX calls to ACWIN-MQ source keys

– Automatic Redirection to Voice Mail

– Multiple Hop Diversion

7.1.9.2 Remote registration of diversion

With the aid of this feature an authorized subscriber in a mixed HiPath 4000/iSDX network can deactivate an immediate / variable call diversion and set a variable call forwarding for another subscriber of the network (i.e. programme and activate it).

Either a subscriber in the system generally has the feature authorization, e.g. the attendant (= one of the privileged subscribers), or he is issued a PIN for the feature, which he has to use together with the feature code and the number of the remote subscriber.

7.1.9.3 Rerouting

As part of this feature, two control mechanisms are implemented in HiPath 4000 (existent in iSDX) that are to become effective in the event of overload (traffic jam) that cannot be bypassed in a transit network node:

- Prevention of internal rerouting: this is to prevent repeated attempts at rerouting in an iSDX/HiPath 4000 network (= internal), since the route, because of the network topology, must keep coming back to the overloaded transit node.

  The overloaded node knows that this is a bottleneck and informs the source node of this. This dispenses with further attempts and immediately issues busy signaling to the calling subscriber; without this function there would be a long lasting reaction time for the subscriber (waiting until a busy signal comes).
- Prevention of external rerouting: external rerouting means an alternative route through the public network or another private network when a transit node is overloaded, under control of the transit node. Control of the acceptable route (using LCR) resides with the originating node.

7.1.9.4 Service Independent Strings

Service-independent strings (SIS) are used in DPNSS1 to transmit extra information in the feature-specific signaling. These can be sent in specified DPNSS1 messages and are used in many features in iSDX.

Features:

1. Internal Rerouting Disabled - IRD

This is defined as the alternative routing of a call via DPNSS on detection of congestion, using the same Destination Address as that used on the call which encountered the congestion.

2. External Rerouting Disabled - ERD

This string may be included to indicate that, on encountering congestion, rerouting external to the DPNSS network shall be disabled.

3. Node Identification (TID)

This string performs Node Identification for display at the attendant. This displayed information will vary as follows:

- If the remote trunk name is included in the received operation, then this will be displayed,
- If the remote trunk name is not included in the received operation but the node access code is, then the node access code will be displayed - If the call originates at the DX trunk, then the node access code that will be displayed at the attendant will be the DX PBX Reference Number - If the call originates at the HiPath 4000 trunk, then the node access code that will be displayed at the DX attendant will be the HiPath 4000 Trunk Address
- If neither the remote trunk name nor the node access code is included in the received operation, then the local trunk name will be displayed.

4. Redirection Control (RDC)

This string is used in cases where a Communication Server other than the originating Communication Server wishes to prevent the originating Communication Server from invoking the redirection service. This string may be sent where there is a possibility that the originating Communication Server may invoke redirection.

5. Route Restriction Class and Call Barring Group (COS)

This SIS divides into three parts and is currently used by DX to control:

- Route restriction for extensions and attendants (Trunk Access Class is carried forward)
- Call Barring for all calling parties (Class of service is sent for extensions and attendants, Trunk main group number is sent for non-DPNSS trunks)
- Facility barring/allowing for extensions, trunks and attendants (Class of service is sent)

6. Password (PASSW)

This string is used by DX for control of remote registration of diversion. The use of PASSW is primarily aimed at services where it is possible to change data held at one Communication Server from another Communication Server.

This string is used in cases where unconditional use of a service is limited to privileged users, but where non-privileged users are entitled to use the service under password control. When such a service is invoked by a non-privileged user, a password is then sent to the required Communication Server for checking.

7. Malicious Call Identification (MCI)

It may be included in a message sent in the backward direction after answer. On receipt of such an indication a Communication Server may, for example:

- record call details for later examination by appropriate authorities including:

– time of call
– identities of calling and connected lines and parties
– dialled number
– Malicious Call Reference (if provided)

7.1.10 CLI (Calling Line Identification) Translation

The CLI (Calling Line Identification) translation correct backwards translations of numbers between CorNet NQ network subscribers and heterogeneous network subscribers (mixed networks e.g. with DPNSS) because of different numbering plans in the network.

For example, when interworking between a CorNet NQ and DPNSS network, a Redirection Party Number is normally sent from the DPNSS network to the HiPath network which is supposed to contain the DPNSS Network Access Number (area code). With CLI translation, a Communication Server (in any DPNSS network) can include its network access number (area code) in the CLI signalling to HiPath . The HiPath receives the correct backwards translation of the called number.

This software enhancement results in a correct Redirection Party Number being formed at the HiPath node receiving the message signalled from the DPNSS-network.

The CLI Translation feature adds the network access number (area code) as required and generates routing information which is recognisable throughout the network.

Subscriber User Interfaces

The subscriber user interface relating to this feature remains unchanged. The subscriber having a terminal with display capability sees the full network number relating to the Redirection Party.

Impacted Features

The following indicates the features which are potentially impacted with the CLI Translation :

1. Call Forward Unconditional,
2. Call Forward on Busy,
3. Call Forward on No Reply

7.2 Basic Package for Exchange Trunks

7.2.1 DSS1 CO-Line Trunk

In many countries, the HiPath 4000 supports the Euro-ISDN trunk protocol DSS1. This allows HiPath 4000 subscribers to use certain services and features (Supplementary Services), depending in the respective public ISDN network.

The DSS1 protocol was adjusted to meet the requirements of the supported countries to

- achieve full technical compatibility to the ISDN trunk and to
- fulfill the conditions for approval and the test regulations.

Definitions

- Euro-ISDN is a technical standard for Europe as defined by ETSI (European Telecommunications Standards Institute) for:
  - the connection/operation of Euro-ISDN stations and terminals to/on the public main station (DSS1 multi device connection),
  - the connection/operation of Communication Servers to/on the public ISDN (DSS1 System connection) and
  - the connection/operation of Euro-ISDN stations to/on the private subscriber line circuit (DSS1 multi device connection behind the Communication Server).
- The ETS standards defined by ETSI for Euro-ISDN are based on the corresponding ITU-T recommendations.
- The D channel signalling protocol (DSS1) is defined in the Euro-ISDN standards for all 3 layers, i.e:
  - the line interfacing equipment (= layer 1) for Basic access (S_0, T_0) and the Primary Rate Access (S_2, T_2),
  - the HDLC procedure for secure D channel signalling (= layer 2),
  - the basic signalling procedure/the basic call, which means the setup and cleardown of connections (= layer 3) and
  - the functional control of the features /Supplementary Services (= layer 3).

7.2.1.1 Services and features

The following table shows the ISDN services and features supported by HiPath 4000 for Euro-ISDN trunks and their possible use in individual countries.

Feature No.	Feature	implement ed for Exch. line T_0/T_2	for the following countries BEL	BRD	DAN	FIN - ATC	FIN - Telecom	FKR	ITA	LUX	NDL	OES	POR	RSA	SWZ	UK - BT	UK - Mercury
	Bearer Services / Note 3																
1	64 kbit/s Unrestricted Bearer Service	+	+	+		+	+	+	+	+	+	+	+	+	+	+	+
2	64 kbit/s Service useable for 3.1 kHz Audio Inform. Transfer	+	+	+		+	+	+	+	+	+	+	+	+	+	+	+
3	64 kbit/s Service useable for Speech	+	+	+		+	+	+	+	+	+	+	+	+	+	+	+
	Circuit-mode Teleservice / Note 4s																
1	Telephony 3.1 kHz	+	+	+		+	+	+	+	+	+	+	+	+	+	+	+
2	Telephony 7 kHz	+	+	+		-	+	+		+		+			+		
3	Teletex 64 K	+	+	+		-	+	-	-	-	-	+			-		
4	Group 4 Telefax	+	+	+		+	+	+	+	+	+	+			+		
	Supplementary services																
1	Direct Dialling In (DDI)	+	+	+		+	+	+	+	+	+	+			+		
3	Calling Line Identification Presentation (CLIP)	+	+	+		+	+	+	+	+	+	+			+		
4	Calling Line Identification Restriction (CLIR)	+	+	+		+	+	+	+	+	+	+			+		
5	Connected Line Identification Presentation (COLP)	+	+	+		-		-	+	+	+	+			+		
6	Connected Line Identification Restriction (COLR)	+	+	+		-		-	+	+	+	+			+		
7	Malicious Call Identification (MCID)	+	+	+		-	+		+	+		+			+		
8	Subaddressing (SUB)	+	+	+		-	+	+	+	+	+	+			+		
9	Advice of Charge (AOC): Charging Information																
9 a	– at Call Set-Up Time (AOC-S)	+	+									+					
9 b	– during the Call (AOC-D)	+	+	+								+					

Table 4 EURO-ISDN country secific Features

Feature No.	Feature	implemented for Exch. line T_0/T_2	BEL	BRD	DAN	FIN - ATC	FIN - Telecom	FKR	ITA	LUX	NDL	OES	POR	RSA	SWZ	UK - BT	UK - Mercury
9c	– at the End of the Call (AOC-E)	+	+	+								+					
10	User to User Signaling Service 1 implicit (UUS1)	+	+	+		-	+	+	+	+		+			+		
11	Call Forwarding Unconditional immediate (CFU) / Note 1)	+	+	+						+		+			+		
12	Call Forwarding Partial Rerouting (CF-PR) / Note 2)	+	+	+						+		+			+		
13	Completion of Calls to Busy Subscriber (CCBS)	+	+	+						+		+			+		
14	User to User Signaling Service 3 (UUS 3)	+		+					+			+			+		

Table 4 EURO-ISDN country secific Features

Note 1)
Permanent call forwarding covers the entire trunk group. All incoming calls are forwarded to the same destination in the public network. With HiPath 4000, CFU can only be used for Telephony 3.1 kHz.

Note 2)
No B channels are seized on the Euro-ISDN trunk if call forwarding per extension is activated. Supported CF-PR variations:

Call forwarding permanent per extension
Call forwarding on busy per extension
Call forwarding on no answer per extension

Note 3)
Other Circuit-mode Bearer Services not listed here are not actively supported by HiPath. The HiPath, however, transmits these services transparently which allows their use for trunk calls.

Note 4)
Other Circuit-mode Teleservices not listed here are not actively supported by HiPath. The HiPath, however, transmits these services transparently which allows their use for trunk calls. No Fallback procedures are supported for the teleservices Telephony 7 kHz and Videotelephony.

Note 5)
Table to be completed.

7.2.1.2 Supplementary Services - Functions

1. Direct Dialing In (DDI)

   Control of DDI connections on the ISDN trunk (Basic Access, Primary Rate Access) as per the Basic Call Procedure defined in ETS 300 102-1.

   - The DDI number in the information element "called party" and the setup message are signalled via the Euro-ISDN trunk to the HiPath 4000.
   - The DDI number is composed according to the "ITU telephony numbering plan (ITU Rec. E 164/E163)"; it always contains the terminal number of the called party and, dependent on the implementation in the network, also the codes from the network.
   - The DDI number can be transmitted either digit-by-digit or ‘en bloc‘.
   - The format of transmitted DDI information is configurable in the HiPath 4000. Parameters: "Type of Number" and "Numbering Plan Identification"

2. **Calling Line Identification Presentation** (CLIP)

   - The option to display the calling party‘s number at the called party is done using the information element "calling party number" in the setup message on the Euro-ISDN trunk. Used e.g. for incoming and outgoing basic trunk calls; procedure for basic call control as per ETS 300 102-1.
   - Supplementary HiPath 4000 -specific definitions for CLIP:
     CLIP is included in the "Station number signalling with open numbering".
     Reason: the composition of the calling party‘s number at the called party is dependent on the HiPath 4000 node to which the called Euro-ISDN party is connected :

     a) the number consists only of parts of the public network if the called party is connected to the HiPath 4000 closest to the CO, or

     b) a combination of public station number + node number of the private HiPath 4000 network, if the called party is connected to the HiPath furthest from the CO.

3. **Calling Line Identification Restriction** (CLIR)

   display of the A station number ("calling party number") at the B station via the ISDN trunk can be suppressed using the parameter ‘Presentation Indicator= Presentation Restricted‘.

   - The A station activates this parameter for outgoing calls. In HiPath 4000 the parameter is carried out using "Display suppression".
   - For incoming trunk calls, Euro-ISDN notifies the called party that the caller is configured with an unlisted number.

   Intended variants: CLIR permanent, CLIR per call and CLIR Suppression per call.

The A party‘s trunk group name and station number are normally displayed for incoming ISDN trunk calls. If CLIR is activated, the display shows ‘*XXX*’. The trunk group name, however, contains no data about the A party which should not be transmitted. Whether the trunk group name should be transmitted, can be configured for every HiPath 4000 individually.

4. **Connected Line Identification Presentation** (COLP)

   When a connection is switched through, the calling party receives the station number of the actual partner (B party) with the CONNECT message in the information element "connected number".

5. **Connected Line Identification Restriction** (COLR)

   The parameter ‘Presentation Indicator=Presentation Restricted‘ in the information element "connected number" is used to suppress the display of the B party‘s number at the connection source (A party).

   For outgoing trunk calls the HiPath 4000 receives this information from the Euro-ISDN in the CONNECT message and transmits it transparently to the station.

6. **Malicious Call Identification** (MCID)

   A station with a trunk call in talk state can initiate the recording of connection-related data within the Euro-ISDN switch by activating this function at the terminal.

   - Supplementary HiPath-specific definitions for MCID:
     For expenditure reasons, MCID is not implemented on HiPath 4000 subscriber line cirucits. Therefore, with HiPath 4000 only analog telephones or the available digital telephones (optiset E, optiPoint 500) can instigate MCID in the CO.

     MCID on the Euro-ISDN trunk: there are two options to configure MCID in the Hipath 4000 in the ISDN switch:

     a) Malicious call identification immediately (MCII)

     b) Malicious call identification on request (MCIR)

     If MCII has been configured, all calls to the Communication Server are automatically registered. Therefore, tracing per call can be initiated from the HiPath 4000 only if MCIR has been configured. The configuration of MCII for HiPath 4000 Communication Servers is not feasible.

     The functional signalling for MCID on a trunk as per ETSI allows many options which have been limited to the necessary minimum for HiPath 4000 implementation:

     a) Implementation only for "Voice".

     b) Due to the new acknowledgement messages from the ISDN switch, no run-on effects to the internal CP interface between station and circuit.

After initiation of MCID in the CO the following acknowledgement messages to the HiPath 4000 are possible (information in the facility IE):

As return result:	invoke ID
As return error value:	not subscribed, not incoming call, suppl. serv.not available
As reject component:	problem tags

7. **Subaddressing** (SUB)

   Besides "MSN", the feature "Subaddressing" is the second method for subaddressing intended in Euro-ISDN to be used at the called subscriber line circuit .

   In addition to the element "called party number" the information element "called party subaddress" can be used to address the called party during basic connection setup. The information element "called party subaddress" allows e.g. :

   a) either terminal selection from several devices with the same services or

   b) process selection on terminals

   on the subscriber line circuit.

   In both cases, the ISDN number in combination with the service IDs (HLC, LLC, BC) is used for global service-specific addressing.

   In contrast to MSN, the "called party subaddress" is a supplement to the ISDN station number and not a part thereof.

   Length of the subaddress information: a maximum of 20 octetts

   The manufacturer defines the encoding in compliance with the ETSI options.

   The "called party subaddress" is transmitted transparently from the calling station to the called station via the network. The subaddress information is therefore only of importance for the stations which support this service and not for the network.

   Note: A "calling party subaddress" (transmitted in the SETUP message) in addition to the "called party subaddress", or a "connected party subaddress" (transmitted in the CONNECT message) is not intended for the service "Subaddressing", but is an option within the services CLIP and COLP.

   Supplementary HiPath 4000-specific definitions for SUB:

   – Validation of the length of the "called party subaddress".

   – The maximum length of the "called party subaddress" can be configured in HiPath 4000. Reason: Network-related limits beneath the maximum length of 20 octetts, specified in the ETS 300 059.

8. **Advice Of Charge** (AOC)

   There are three AOC variants as defined by ETSI:

a) AOC at call set-up time, permanent and per call (AOC-S)

b) AOC cumulative during the call, permanent and per call (AOC-D)

c) AOC at the end of the call, permanent and per call (AOC-E)

With AOC-S, the call charge rate (= call charge unit per call charge pulse) is transmitted and not the call charge information (call charge units or amounts). The station receives the call charge rate during connection setup or at the latests with the CONNECT message from the switch.

Additional option: If the call charge rate changes during the call, the station is notified of the new rate.

With AOC-D the accrued call charge sum is transmitted during the active call as call charge units or amounts with currency denomination.

Time scale for the transmission: Either in fixed intervals which can be configured in the network, or if the call charge total has reached a previously configured amount.

The station is notified if a connection is free of charge.

With AOC-E, the total amount of the accrued call charges (call charge units or amounts with currency denomination) is transmitted with the first release message at the end of the call.

AOC-E is included in AOC-D.

All three AOC variants can be activated

– automatically for all connections or

– for every connection separately or

– on request by the station.

All three AOC variants are activated/deactivated by functional signalling using the "facility information elements" in the SETUP message at the start of a connection.

Supplementary HiPath-specific definitions for AOC:

Basic requirement: Integration into the existing call charge concept as easy as possible. A solution which meets this basic requirement has the following characteristics:

a) Reduction of AOC variants to be implemented:

   HiPath 4000 supports AOC-S only for DSS1 trunks, not for tie trunks (CorNet NQ, QSIG) or the DSS1 connections behind the HiPath 4000 (DSS1-MgAs, DSS1-AnlAs). In certain countries AOC-S is used to transmit call charge rates required for correct call detail recording.

   If the HiPath 4000 is to evaluate call charge rates on trunks, AOC-S (permanent) must be configured in the ISDN switch.

AOC-E evaluation on trunks only in those countries, where call charge transmission with AOC-D at the end of a call is not possible.

b) Flexible adjustment to call charge units or amounts.

Depending on the country, AOC-D/E allows the transmission of either call charge units or amounts. HiPath 4000 provides a solution for both possibilities.

c) Implementation of parts of "GESP" from the ETSI standard: this means, that the range of AOC-D functions at the Euro-ISDN subscriber line circuit correspond mainly to the current GESP functionality. AOC-D is

– permanently available for every Euro-ISDN station; it cannot be activated/deactivated from the station

– valid for all calls.

d) HiPath 4000 does not support any "free running" call charges, i.e. no AOC-E messages transmitted by the public ISDN switch on the ISDN trunk without an assigned existing B channel connection.

Such "free running" call charge messages signal incoming trunk call charges which are charged to the Communication Server provider, if this trunk call is forwarded (using CFU or CF-PR) to a station in the public network (for the second path from the Communication Server to the forwarding destination). For this kind of call forwarding no B channels are seized on the ISDN trunk.

Result:
If CFU and CF-PR are used on ISDN trunks, the call charges calculated by HiPath will not correspond to the call charges calculated by the Telekom.

9. **User to User Signalling Service 1 implicit** (UUS1i)

The capacity of the data transmitted in the ’User-User Information Element’ of the SETUP message is extendedfrom 128 up to 131 octets inHiPath 4000, as defined in 1TR67.

10. **Call Forwarding Unconditional** (CFU)

IF CFU is active, all incoming calls from the public ISDN (permanent) are forwarded to a number other than that dialed in the public ISDN. Call forwarding is carried out directly in the ISDN switch. Therefore, no B channels are seized on the Euro-ISDN trunk.

Call forwarding unconditional covers the entire trunk group, i.e. all incoming calls for this trunk group are forwarded to the same forwarding destination in the public ISDN.

CFU can be used independent of services.

Authorized HiPath stations can enter the forwarding destination and activate/ deactivate CFU.

11. **Call Forwarding Partial Rerouting** (CF-PR)

    CF-PR allows call forwarding in the ISDN trunk per individual extension. No B channels are seized on the Euro-ISDN trunk.

    The following CF-PR variants are supported:

    – Call forwarding permanent per extension

    – Call forwarding on busy per extension

    – Call forwarding on no answer per extension

12. **Completion of Calls to Busy Subscriber** (CCBS)

    Automatic number redial when the call meets a busy station, and the station changes from busy to idle.

    The implementation of CCBS on the ISDN trunk allows a HiPath station to initiate automatic callback for outgoing trunk calls if the called party in the public network is busy. As soon as the called party becomes idle, the HiPath station receives a callback. If the callback is answered, the public ISDN automatically establishes the connection.

    CCBS can also be used for incoming trunk calls, if the called HiPath station is busy and the calling party in the public ISDN initiates automatic callback. As soon as the HiPath station becomes idle, the station in the public ISDN receives a callback. If the callback is answered, the public ISDN automatically sets up the connection.

13. **UUS 3 Request During Calling Phase** (UUS3)

    The UUS3 feature enables a user to exchange a certain amount of information with the other party during an active call. During this process, UUS3 is explicitly requested by means of indicators, confirmation by the remote party is also required.

    UUS3 is a guaranteed bearer service, i.e. if the feature is not supported, the user making the transmission is informed.

    UUS3 can be requested by the calling party during call setup or during an active call. The called party can <u>only</u> request UUS3 during an active call.

    UUS3 can be requested as "essential" or "non essential". "Essential" signifies that the basic call must be rejected if the requested feature is not supported. In the case of a request during an active call, UUS3 can only be selected as "non essential".

    In BRD the public network limits the number of User Information messages per direction to 16 messages per 10 sec.

    UUS3 is used in conjunction with all circuit-switched services that are supported on DSS1 CO-line HiPath 4000.

7.2.1.3 Notifications

The notifications (advisory messages) on different DSS1 SS (ECT, HOLD, CD, TP, 3PTY, CW) are transferred and generated.

DSS1 terminals that are connected to the DSS1 passive bus or ISDN systems connected to the HiPath 4000 DSS1 system connection can, in the context of the supplementary services supported by HiPath, send and receive notifications to and from remote users.

In the case of outgoing external traffic, all notifications that are possible in the context of the supplementary services currently defined in the standards by ETSI and realised in the public ISDN (see Table 1) are transparently routed over the DSS1 CO line to the public ISDN.

In the case of incoming external traffic, all notifications received over the DSS1 CO line are transparently routed to the DSS1 terminals connected to HiPath or to the DSS 1system connection.

It is also possible to exchange notifications if the DSS1 CO line and the DSS1 passive bus or DSS1 system connection are connected to two different HiPath 4000 systems that are linked together over CorNet NQ.

It is possible to utilise notifications in conjunction with all circuit switched services that are supported on the DSS1 CO line.

Unattached notifications that do not belong to any existing trunk call are not supported.

7.3 Extended Networking Functions

7.3.1 DSS1 System connection

The public Euro-ISDN offers two ISDN connection types, which vary in detail:

1. the DSS1 multiple device connection (DTAG designation) for the connection of Euro-ISDN terminals.
2. the DSS1 System connection (DTAG designation) for the connection of Euro-ISDN PABX.

The current market offers a variety of Euro-ISDN terminals and PABXs for both connection types, which customers could use on HiPath systems. The market demands that terminals and PABXs, when connected to HiPath 4000 systems, allow the use of the same services and supplementary services as provided by the public ISDN.

The public ISDN offers two interface variants for DSS1 System connections:

1. as T_0 interface (international designation) or S_0 interface (DTAG designation) in point-to-point configuration.
   Other designation: Basic Access
2. as T_2 interface (international designation) or S_{2M}interface (DTAG designation) in point-to-point configuration.
   Alternative designation: Primary Rate Access

Figure 28 provides an overview of ISDN interfaces both in the public ISDN and HiPath 4000 . The DSS1 System connection behind the HiPath 4000 described in this section is represented as a "bold" line.

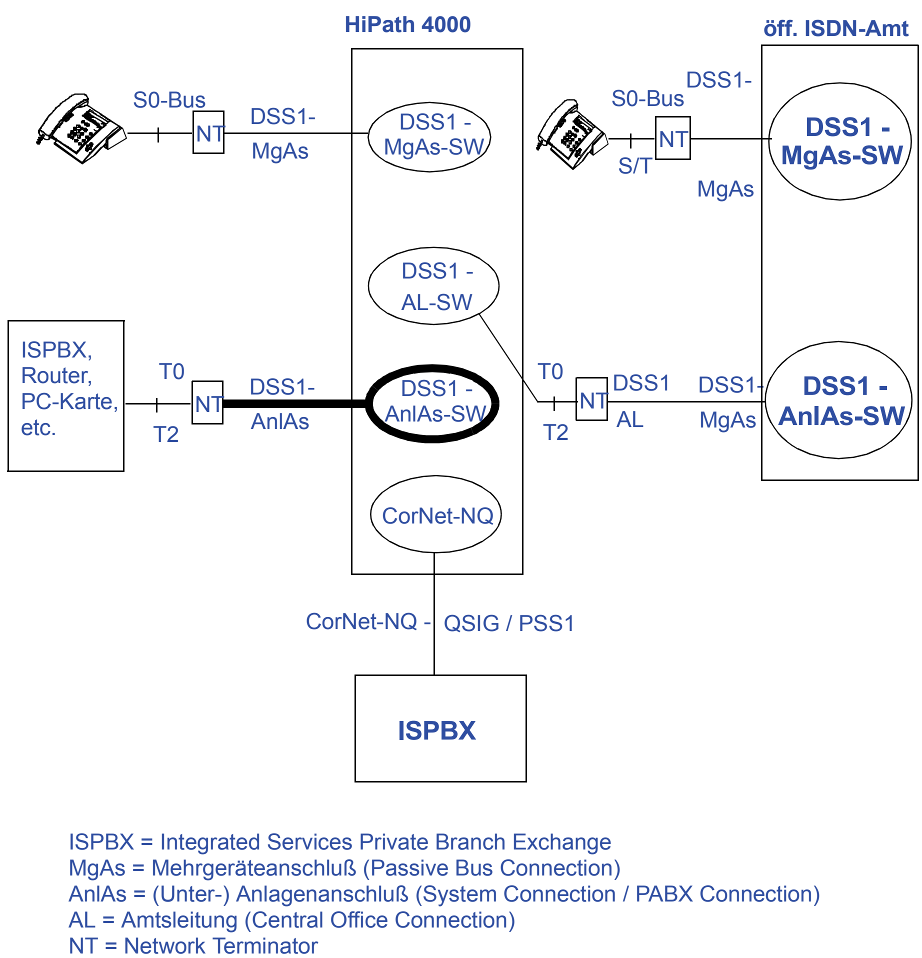


Figure 28 Overview of the ISDN interfaces in the public ISDN and on HiPath 4000

7.3.1.1 Configurations

Figure 29 provides configuration overview for DSS1 System connections supported by HiPath 4000. The line modules and network terminators needed to implement these configurations are represented.

With all configurations supported by HiPath, the users of DSS1 System connections have access to the same services and features.

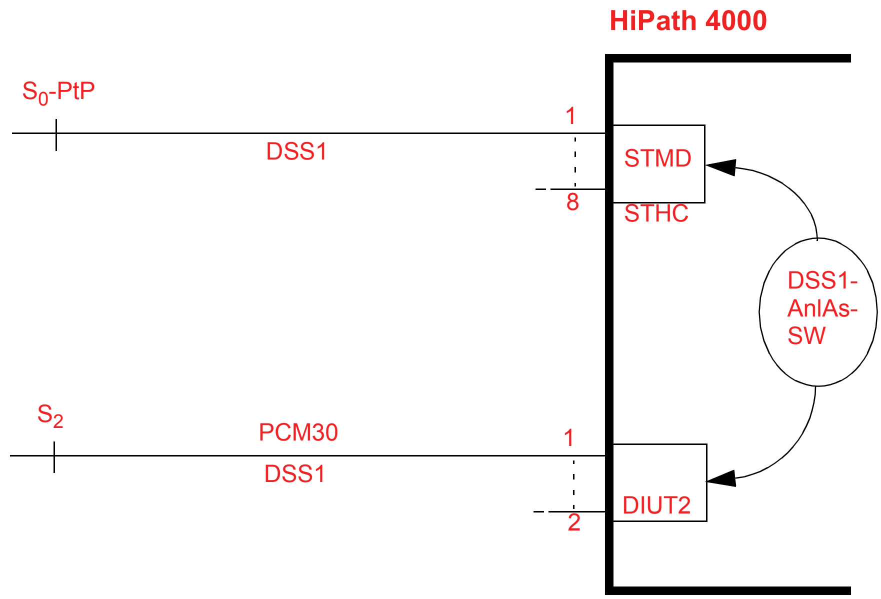


Figure 29 Components for the implementation of DSS1 System connections

7.3.1.2 Country-specific variants

Individual countries use specific variations of the DSS1 protocol, which is standardized by ETSI, for the public ISDN. In HiPath 4000, the DSS1 System connection is supported according to the DTAG, DES, ITL, SWZ specifications.

Further country-specific adjustments of the DSS1 System connection can be implemented, if required, to achieve complete technical compatibility with the Euro-ISDN terminals available in the respective country. Generally, these adjustments are independent from the HiPath version planning.

7.3.1.3 Services and supplementary services

The following sections describe the services and supplementary services standardized for Euro-ISDN and supported by the HiPath 4000 on the DSS1 System connection.

7.3.1.4 Bearer Services and Teleservices

On the DSS1 System connection, HiPath 4000 supports the connection setup for all Circuit-mode Bearer Services and Circuit-mode Teleservices used in public Euro-ISDN.

The following Circuit-mode Bearer Services are supported:

- 64 kbps Unrestricted Bearer Service
  64 kbit/s Bearer Service
- 64 kbps Service usuable for 3.1 kHz Audio Information Transfer
  3,1 kHz Bearer Service
- 64 kbps Service usuable for Speech Information Transfer
  Speech Transfer Service

HiPath transfers all other Circuit-mode Bearer Services in transparent mode.

The following Circuit-mode Teleservices are supported:

- Telephony 3.1 kHz
- Telephony 7 kHz
- Telefax Group 4

HiPath 4000 transfers all other Circuit-mode Teleservices in transparent mode. Fallback procedures for the teleservices Telephony 7 kHz and Videotelephony are not supported.

7.3.1.5 Supplementary Services

The following DSS1 Supplementary Services for the DSS1 System connection are supported since HiPath 4000

DDI	Direct Dialing In
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
COLP	Connected Line Identification Presentation
COLR	Connected Line Identification Restriction
MCID	Malicious Call Identification
SUB	Sub-Addressing
UUS1i	User-to-User Signalling Service 1 implicit
AOC-D	Advice of Charge during the Call
AOC-E	Advice of Charge at the End of the Call

HiPath 4000 supports the following DSS-1 features for the system connection (AnlAs):

CF-PR	Call Forwarding Partial Rerouting (CFU, CFB, CFNR)
CCBS	Completion of Calls to Busy Subscriber
UUS3:	User to User Signalling

The following sections contain a brief description of Supplementary Services available forHiPath 4000. For a detailed description of the implemented Supplementary Services with options and exceptions see the PICS forms (PICS = Protocol Implementation Conformance Statements) defined by ETSI and filled in for the HiPath 4000.

1. **Direct Dialing In** (DDI)

   DDI allows direct calls to stations within an Euro-ISDN system (e.g. a PABX) which is directly connected to the DSS1 System connection of a HiPath 4000.

   The numbering plan as per ITU-T recommendation E.164 applies for the connection setup via the DSS1 System connection. The dialing information can be transmitted either ‘en bloc‘ or digit by digit (overlap).

2. **Calling Line Identification Presentation** (CLIP)

   CLIP allows signalling and display of the calling party‘s number at the called party.

3. **Calling Line Identification Restriction** (CLIR)

   CLIR allows a calling party to suppress the display of his number and, if used, a subaddress (SUB) at the called party.

   Suppression follows the ETSI standard and can be activated either in permanent or temporary mode, i.e. on request of the calling party. The HiPath 4000 supports **both** modes.

   As per ETSI standard there are the following default procedures for the temporary mode:

   – Default value is "Presentation restricted". A user must explicitly signal, if the station number is not to be suppressed for a certain call.

   – Default value is "Presentation not restricted". A user must explicitly signal, if the station number is to be suppressed for a certain call.

   On HiPath 4000 System connections CLIP/CLIR can be configured either as "CLIP permanent", "CLIP default, CLIR per call " CLIR permanent" or "CLIR default, CLIP per call" .

   HiPath 4000 supports the CLIR option "Override Category". Although a calling station has CLIR, the station number and, if used, the subaddress are displayed at the called station, if the called party has the feature "Override Category".

4. **Connected Line Identification Presentation** (COLP)

   COLP allows the calling station to have the number of the called party displayed.

5. **Connected Line Identification Restriction** (COLR)

   COLR allows a called station to have the display of his station number and, if used, his subaddress (SUB) suppressed at the calling party.

Suppression follows the ETSI standard and can be activated either in permanent mode or in temporary mode, i.e. on request of the called party. The HiPath 4000 supports **both** modes.

On HiPath 4000 System connections COLP/COLR can be configured either as "COLP permanent", "COLP default, COLR per call " COLR permanent" or "COLR default, COLP per call".

The HiPath 4000 System connection supports the COLR option "Override Category". If the calling party has this option activated, the station number and, if used, the subaddress are displayed,although the called station has COLR.

6. **Malicious Call Identification** (MCID)

   MCID allows the identification of malicious callers. In the **public ISDN**, the following MCID variants can be administrated for the DSS1 System connection:

   a) Malicious call identification immediately (MCII):
   Every incoming call to a Euro ISDN PABX directly connected to a DSS1 System connection is automatically identified.

   b) Malicious call identification on request (MCIR):
   Incoming calls are only identified when requested by the called party. Identification can be requested during, or shortly after a call.

   With HiPath 4000 only incoming calls from the public ISDN can be traced (i.e. calls from stations within a HiPath system cannot be traced). The connection-related identification data is recorded in the ISDN CO. If MCII is configured in the ISDN CO for the DSS1 trunk, *all* incoming trunk calls are identified. If only incoming trunk calls for a certain DSS1 System connection on HiPath are to be identified, MCIR must be configured in the ISDN CO.

   With HiPath 4000, MCID can only be used for voice services.

7. **Sub-Addressing** (SUB)

   SUB allows both the calling and the called station to signal a subaddress in addition to the station number. The subaddress is then used for the selection of certain stations or for process control in DTEs. The subaddress is transmitted in transparent mode.

   The maximum length of the subaddress is 20 octetts (maximum length of the information element is 23 octetts).

8. **User to User Signalling Service 1 implicit** (UUS1i)

   UUS1i allows the D channel to be used for transparent bidirectional transmission of a limited quantity of user data between two stations during call setup and call cleardown.

   The capacity of the data transmitted in the 'User-User Information Element' of the SETUP message is extended up to 131 octets in HiPath 4000, as defined in 1TR67.

9. **Advice Of Charge during the Call** (AOC-D))

   If AOC-D is activated, the HiPath 4000 transmits the accumulated call charges via the DSS1 System connection to the Euro-ISDN system in defined intervals. The call charges are transmitted either as call charge units or amounts with currency designation during the active call. Both call charges received from the public ISDN and call charges generated by the HiPath are transmitted.

   HiPath 4000 provides permanent call charge information via the DSS1 PABX conenction. Activation/deactivation of AOC-D from the Euro-ISDN system is not supported.

10. **Advice Of Charge at the End of the Call** (AOC-E)

    If AOC-E is activated the HiPath tansmits the total amount of accumulated call charges (call charge units or amounts with currency designation) at the end of the call via the DSS1 System connection. Both call charges received from the public ISDN and call charges generated by the HiPath are transmitted.

    HiPath 4000 provides permanent call charge information via the DSS1 PABX conenction. Activation/deactivation of AOC-E from the Euro-ISDN system and its stations is not supported.

    The HiPath 4000 does not support "Free running" call charges, i.e. AOC-E messages transmitted without any assigned B channels on the DSS1 System connection.

11. **Call Forwarding Partial Rerouting** (CF-PR)

    The CF-PR feature means, from the viewpoint of the user, the extension-specific forwarding of external calls without seizure of B-channels on the DSS1 system connection .

    If a user of an exchange connected to a DSS1 system connection has activated call forwarding, an incoming external call is forwarded by the HiPath 4000 without B-channels being seized on the system connection . This is particularly important for small system connection trunk groups, or if a large number of users utilise external call forwarding simultaneously.

    An incoming external call is provided to the ISDN exchange over the D-channel of the DSS1 system connection. If the party called has activated forwarding of external calls, this is signalled back to the HiPath together with the forwarding destination. HiPath 4000 then forwards this call to the required forwarding destination within the HiPath 4000 network or the public ISDN.

    CF-PR is realised for the following call forwarding variants that the user can utilise on the exchange:

    – External call forwarding unconditional (CFU)

    – External call forwarding busy (CFB)

    – External call forwarding no reply (CFNR)

– External call forwarding during the calling phase (Call Deflection)

CF-PR is used in conjunction with all circuit-switched services that are supported on the DSS1 system connection.

The programming, querying and deleting of call forwarding **(CFU, CFB, CFNR) for the entire** user side of the system connection (AnlAs**)** .

In future, the public ISDN will signal the charges incurred for a call forwarded via CF-PR, and that are to met by the exchange operator, by means of "unattached" AOC messages (i.e. AOC messages without B-channel allocation) over the DSS1system connection. This is prerequisite for the call charges calculated by the exchange matching the PTT invoice. These unallocated AOC messages are not supported.

12. **Completion of Calls to Busy Subscriber** (CCBS)

    The CCBS feature on the DSS1 system connection makes it possible for the calling party who encounters a busy destination to store a job for automatic callback. Once the destination is free, the party who initiated the callback is called and a new call to the destination automatically started when the calling party answers.

    CCBS affects the DSS1 system connection in both traffic directions.

    CCBS is used in conjunction with all circuit-switched services that are supported on the DSS1 system connection .

13. **User-To-User Signalling Service 3** (UUS3)

    The UUS3 feature enables a user to exchange a certain amount of information with the other party during an active call. During this process, UUS3 is explicitly requested by means of indicators, confirmation by the remote party is also required.

    UUS3 is a guaranteed bearer service, i.e. if the feature is not supported, the user making the transmission is informed.

    UUS3 can be requested by the calling party during call setup or during an active call. The called party can only request UUS3 during an active call.

    UUS3 can be requested as "essential" or "non essential". "Essential" signifies that the basic call must be rejected if the requested feature is not supported. In the case of a request during an active call, UUS3 can only be selected as "non essential".

    In Germany the public network limits the number of User Information messages per direction to 16 messages per 10 sec.

    UUS3 is used in conjunction with all circuit-switched services that are supported on DSS1 system connection HiPath 4000.

7.3.1.6 Interworking

Euro-ISDN systems linked to the DSS1 System connection of a HiPath 4000 can establish connections to terminals or other systems connected to the same HiPath. They can also establish connections within a homogenous or heterogenous HiPath network, within the public ISDN or within the public analog network. All services supported by HiPath 4000 for the DSS1 System connection can be used, provided they are also supported by the partner systems and the networking components.

An interworking function implemented in HiPath 4000 allows the combination of Euro-ISDN systems linked to a DSS1 System connection and analog stations, or stations with HiPath system interfaces, as far as technically possible.

For HiPath 4000, interworking is impelemented to the following subscriber, tie and trunk lines: (a subset of the available connection types are exhibited in Figure 30):

- DSS1 trunks
- Analog trunks
- Trunks with CAS signalling
- DSS1 System connections
- CorNet-NQ tie lines
- QSIG-V1 tie lines (if relevant for the Supplementary Services supported on the DSS1 System connection)
- QSIG-V2 tie lines (if relevant for the Supplementary Services supported on the DSS1 System connection)
- PSS1 tie lines (if relevant for the Supplementary Services supported on the DSS1 System connection)
- DPNSS1 tie lines (via CDG)
- CorNet-TS subscriber line circuits
- CorNet-TR line circuts for cordless subscriber
- Attendant console connections
- a/b subscriber line circuits

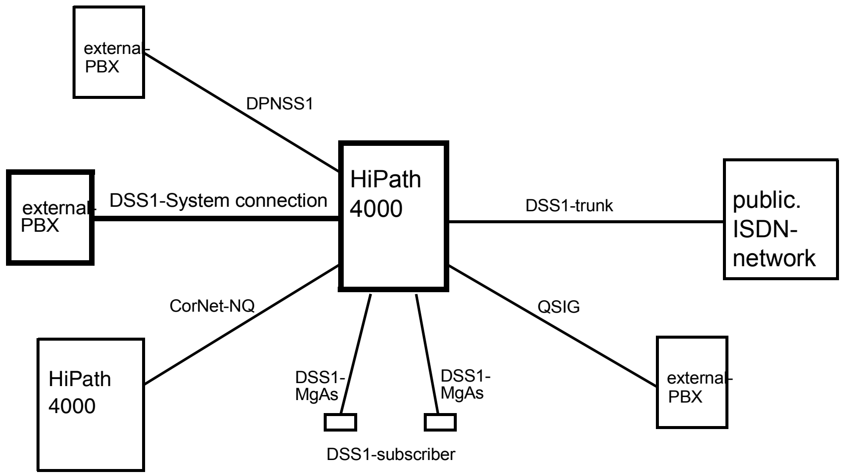


Figure 30 Interworking DSS1 System connection

7.3.1.7 Certifications

To maintain the reliability and high quality in HiPath 4000 networks, external Communication Servers must be certified before they can be connected to the HiPath 4000 QSIG/PSS1 interface. The certifications have to be done by a Siemens certification test laboratory. Normally, protocol tests in the laboratory and field trials are required.

Already certified foreign products must be re-certified in the following cases:

- upgrading of a HiPath 4000 to a new software version.
- if a new version of the foreing product is used.

8 Routing

Least cost routing is an alternate routing system which enables calls to be switched through selected, cost-effective routes outside the PABX.

8.1 Routing

LCR allows calls to be re-routed according to the route code dialled.

The alternate routes can be:

– first route element in private network

– second route element in Mercury network

– third route element in public network.

For network-wide features, which can only be transmitted via CorNet-NQ lines, the LCR system can select routes via CorNet lines exclusively.

If cost-effective routing is dependent on the source node of the call, this can be achieved by configuring alternate dial plan groups or by means of digit prefixing. The alternate dial plan groups can be used if the various source nodes are linked via different trunk groups. If route selection in a transit node is dependent on the source node, and the source nodes all use the same trunk group links, different route codes must be set individually in each source node by means of digit prefixing. The routes can then be selected depending on the route code prefixed. The route code prefixes are not transparent to the network users.

8.2 Rerouting

If a call is switched via an alternate LCR route, and an unexpected "all-trunks-busy" state is encountered en route, the node in which this occurs signals this state to the previous node. This allows the previous node to re-route the call by seizing a different exchange line or tie-line. This is defined as "LCR with re-routing". Each node which receives an "all-trunks-busy" signal from the next node in the route will attempt to re-route the call before transferring it back to the previous node.

Alternate LCR with re-routing ignores all route variants already considered by the LCR.

The AMO command allows different re-routing options to be defined for each node in the route, i.e. whether re-routing is carried out in all nodes, in specific nodes or only in the source node.

8.3 CO Breakout

Whenever a PABX user dials an exchange number outside the local exchange area (i.e. the call is not charged as a local call), the area code dialled by the user is checked by the system to determine whether a private network node exists in the remote area. If this is the case, by replacing the area code digits , the remote private network node will be used as a transit node in order to access its local exchange lines. This means that the call will be charged at local call rates as opposed to the more expensive long distance rate.

When the user dials the exchange code, he or she will hear the exchange dialling tone, regardless of whether the exchange access is re-routed or not.

The parameters of the simulated exchange dialling tone are AMO-defined.

8.4 Out-Dial Rules

Dialling information transmission is controlled by dialling sequences or "out-dial rules" which can be set via AMO.
An out-dial rule consists of a series of commands, e.g. "transmit x digits", "pause", "wait for dialling tone", "change signalling type " etc.
The commands are executed in sequence.
Out-dial rules are assigned for each route.

Each route element is assigned an out-dial rule, which contains control commands for dialling information transmission.

These commands are:

ECHO ALL	transmit all remaining digits	
ECHO	transmit part of the dialled digit sequence	
	Parameter:	Definition of field
OUTPULSE	transmit fixed digit sequence	
	Parameter:	digit sequence (max. 22, e.g. MCL access code)
PAUSE	delay dialled digit transmission	
	Parameter:	delay time
DETECT	dialling tone detection (not for MCL dialling tone)	
	Parameter:	Dial tone marker, timer
TOGGLE	switch between DTMF signals and original signalling type	
AUTHORISATION SEND	transmit an authorisation code from the authorisation code table	
	Parameter:	index of authorisation code for max. 4 users per ITR group
CALL SERIAL SEND	transmit value of call serial counter	

	Parameter:	index of call serial counter for max. 4 users per ITR group
END	end of command sequence	

The out-dial rules can be set via AMO. They are assigned to the route elements when configuring the LCR system.

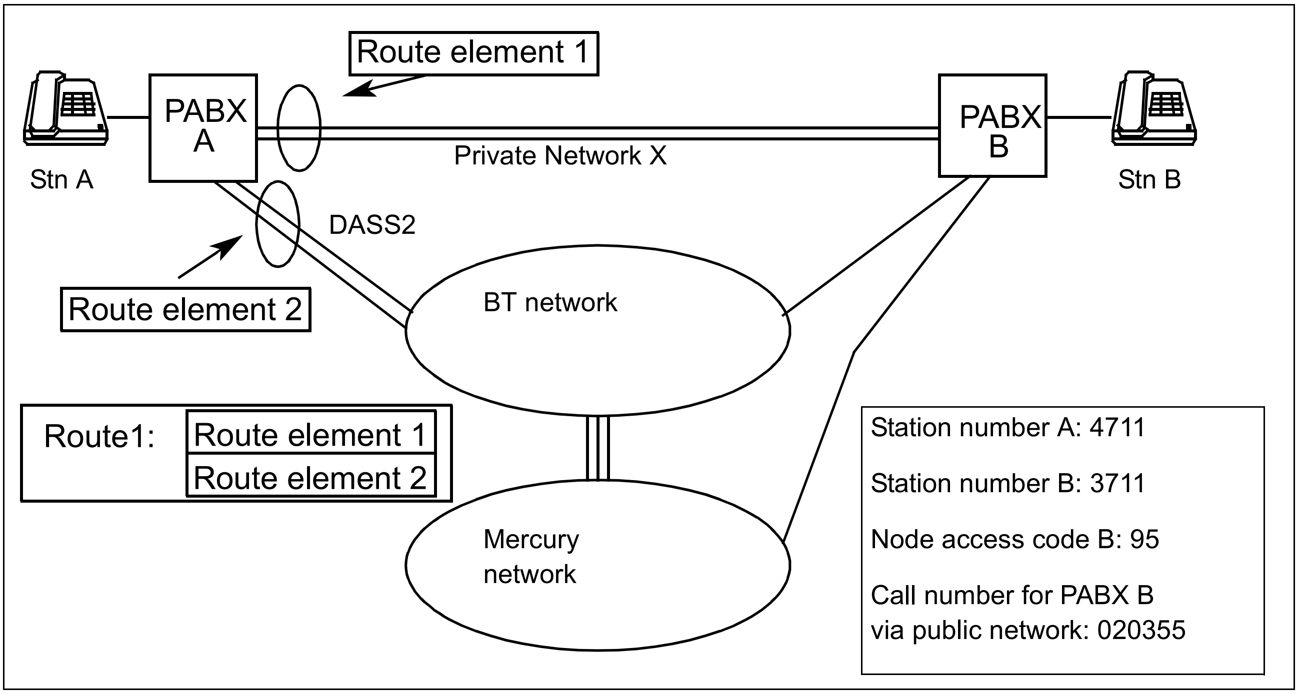


Figure 31 Example for Out-Dial-Rules

The private network X shown here has an open numbering plan, i.e. user A must dial 95 3711 in order to reach user B.

The digit sequence 95 3 is assigned to route 1 in the LCR dialling plan, i.e. the LCR system will set up the call via route 1 each time a dialled number begins with the digit sequence 95 3.

Route 1 consists of 2 route elements:

The first route element leads to system B via the private network X.

The second route element leads via the British Telecom network to the Mercury network , and from there to PABX B. In order to set up a connection via the second route element, the digit sequence dialled by the user must be modified before transmission.

The route element is assigned an out-dial rule, which contains control commands for dialling information transmission. In this example, the out-dial rule consists of the following commands, which are executed in sequence:

OUT PULSE	3, 131:	transmit the three-digit MCL access code (131)
PAUSE	3:	delay three seconds to compensate for dialling tone detection
TOGGLE:		switch to DTMF signalling on B-channel
AUTHORISATION SEND	3:	transmit code 3 in authorisation code table
OUT PULSE	6, 020 355:	Transmit the 6-digit call number of PABX B to the public network exchange
ECHO:		transmit internal station number of user B as dialled by user A (3711)
END		

- Dialling plan

  Route selection is dependent on the entire digit sequence dialled by the user, and not simply the dialled exchange code or route code.

  Route-codes always define service-specific routes (e.g. voice connections, fax , data connections).

  For "voice" and "Data" service connections, route selection can be carried out according to the LCR dialling plan.

  The dialling plan is configured via AMO.

- LCR class of service

  Users are assigned LCR authorisations (classmarks) in the form of an LCR class of service. The LCR COS contains authorisations for various LCR functions, e.g. access authorisations for route elements, for queuing, schedules, route advance timer etc.

8.5 Route Optimisation

Route optimisation is carried out after connection transfer following consultation hold or after connecting by an attendant.

- A connection route between two parties is optimised if the connection was set up by transfer from consultation hold, **and neither** of the two parties (A and C) connected are in the same node as the original called party (B).

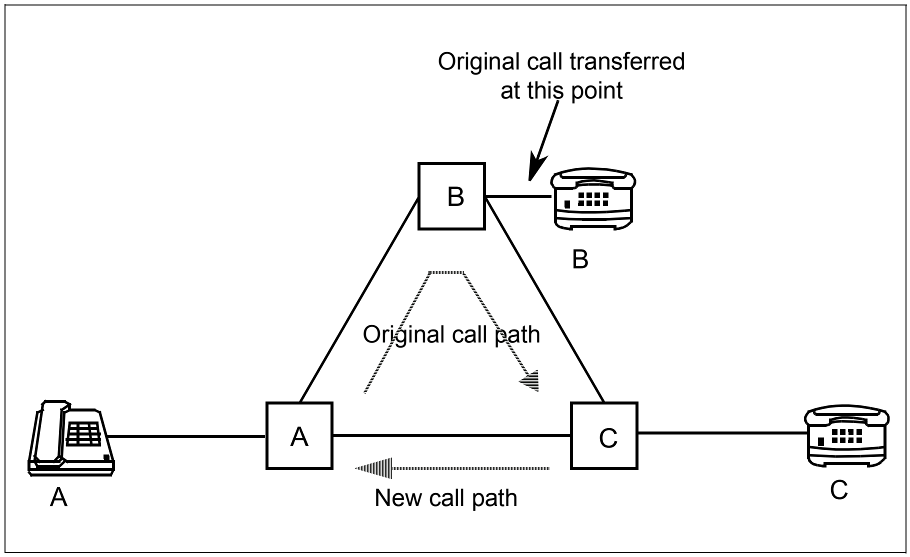


Figure 32 *Example of Route Optimisation*

If the caller (A) is in the same node as the original called party (B), no route optimisation request message is initiated.

If the call transfer recipient (C) is in the same node as the transfer initiator (B), the route optimisation request message is rejected by user C, and route optimisation is aborted.

- A connection route between two parties is optimised if the connection was set up via an attendant (B), **and neither** of the two parties (A and C) connected are in the same node as the attendant console.
- A connection route between two parties is not optimised until **after** the connection has been completely transferred (is in talking state), i.e.:
  - **after** call transfer by consultation call
  - **after** connection by the attendant
  - **after** transfer recipient answers held call
  - **after** called party answers call connected by attendant
- Route optimisation is only carried out after initiation by the above events. Route optimisation does not optimise routes set up via the LCR.
- The optimised route can be a route in **another node or other nodes** of a network, **or** an **internal** route.
- Route optimisation is carried out according to the LCR plan of the node in which user C (transfer recipient/attendant connect recipient) is located.

If the LCR plan envisages **re-routing via the public network exchange**, this is **not** carried out for route optimisation. This is in order to avoid incurring charges on a previously non-chargeable connection (also due to restricted signalling possibilities of public network ISDN exchanges).

Limiting route optimisation to primary routes is not acceptable, since primary routes in private networks often have a high traffic overflow rate. During periods of high traffic, the chances of achieving an optimised route over several nodes would be small.

Due to the number of routes configured in a networked system, with all their complex alternate routing and re-routing rules, creating parallel routes specially for route optimisation is also not acceptable for reasons of serviceability.

For this reason, route optimisation is carried out according to the LCR or alternate routing plan which would be used if user C were to call user A directly, **except** when this plan envisages **re-routing to the public network exchange**. Route optimisation can also be configured to use the **primary route trunk** group exclusively.

- If route optimisation is not carried out, **the original connection route remains intact**. Route optimisation is not carried out if the function is blocked via AMO for a particular route, if no optimised route can be found (all trunks busy), or if CorNet NQ signalling (including DPNSS1) is not possible over the entire route. Route optimisation is only carried out if a route can be found which allows CorNet NQ signalling throughout and the route is available and can be switched.
- Route optimisation is carried out in network areas which use the **CorNet NQ protocol.**

  Route optimisation is carried out between **two user end nodes and/or exchange transit nodes within** the CorNet NQ area.

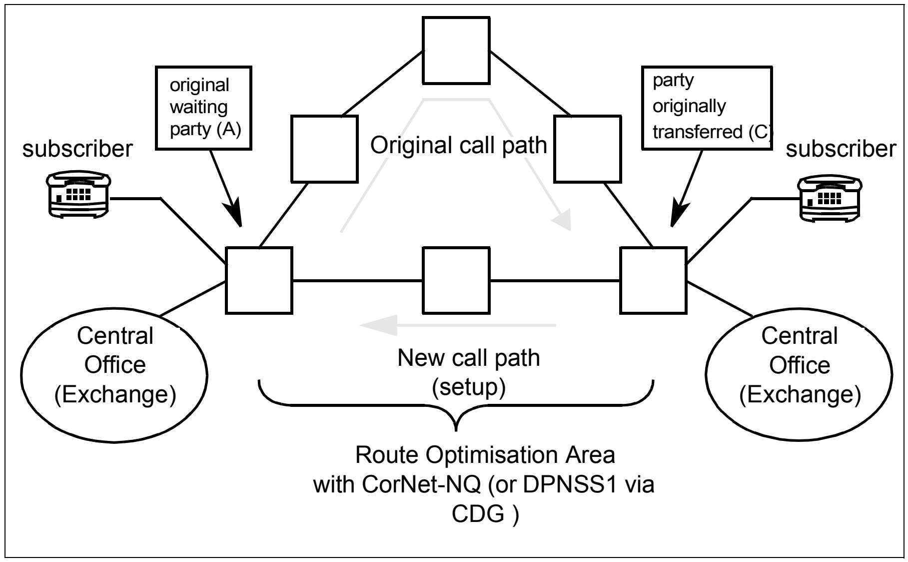


Figure 33 Area of Route Optimisation

- Route optimisation is always initiated by means of a route optimisation request from user A to user C. The optimised route is then set up in the reverse direction, i.e. from user C to user A.
- This call setup order, which also corresponds to the DPNSS1 setup rules, also applies for route optimisation between two transit nodes, i.e. if users A and C are trunk circuits or exchange circuits.
  Route optimisation between two external line circuits assumes that the external lines can be addressed by the route optimisation setup from anywhere in the network (i.e. assumes that the node access codes are unambiguous throughout the network).
- Route optimisation must be enabled or disabled by AMO for each route in each node within an area of route optimisation.
  Whether route optimisation is enabled or disabled is not checked by the node from which the route optimisation request is sent, but in the node from which the optimised route is to be set up.
- Route optimisation between two users is implemented via S0/S2 lines with CorNet NQsignalling, and via the CDG gateway to DPNSS1 networks. The CDG converts the CorNet-NQ signalling for route optimisation to DPNSS1 signalling.
- If the route optimisation setup encounters an "all-trunks-busy" state, alternate re-routing is not carried out, i.e. in this case the route optimisation setup is not repeated.

8.6 Alternate Re-Routing

- If a call is switched via an LCR route, and an unexpected "all-trunks-busy" state is encountered en route, the node in which this occurs signals this state to the previous node. This allows the previous node to re-route the call by seizing a different exchange line or tie-line.
  - To achieve this, the transit node with the "all-trunks-busy" state signals the ATB state to the previous node and performs a backward release on the connection.
  - The previous node can now attempt to re-route the call via a different transit node.
  - Each node along a route can re-route a connection setup received back due to ATB in the next node, including the source node.
- Alternate re-routing is an important requirement of the LCR. However, alternate re-routing can also be used in private networks with LCR.
- There are two variants for alternate re-routing: either alternate routes can be used (i.e. via different trunk group combinations), or the same route can be used but with a modified digit sequence (different route access code for same trunk group combination).
  - Re-routing for calls encountering "ATB" states en route must be set via AMO for each route and in each node.
  - The AMO command allows different re-routing options to be defined for each node in the route, i.e. whether re-routing is carried out in all nodes, in specific nodes or only in the source node.
- Re-routing is carried out whenever one of the following situations is detected (disconnect/release causes):
  - No dial plan entry exists for dialled number in partner system (no route to destination)
  - B-channel negotiation fails (channel unacceptable, requested circuit/ channel not available)
  - Circuit defective/not in operation (circuit out of order)
  - No valid/free circuit available (no circuit/channel available)
  - Network failure (network out of order)
  - Insufficient information for connection setup in remote system (temporary failure, resource unavailable)
  - Resource shortage in partner system (switching equipment congestion)
  - No authorisation for dialled destination (outgoing call barred)

– Line exists, but does not meet connection requirements (service or option not available)

– Requested channel does not exist (identified channel does not exist)

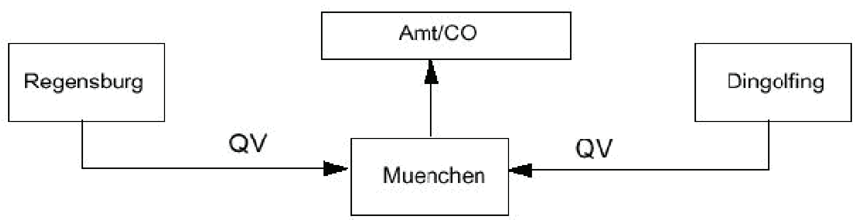
– No response within specified time (recovery on timer expiry).

8.7 Source Dependent Routing

The feature enables to choose a different LCR route depending on originating (source) node. In HiPath 4000 this requirement is translated to a source/destination controlled modification of Least Cost Routing Access Rights (LCR-AR).

Examples:

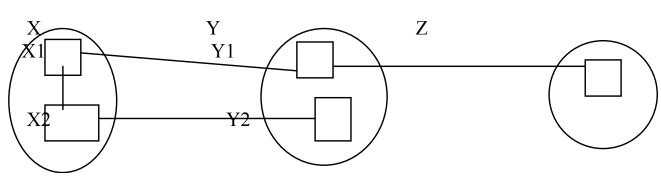
1.



In the example, shown above, all connections originated at Dingolfing are prevented in Munich transit node to be routed via CO overflow. Connections being originated at Regensburg however are allowed to be routed in Munich via CO overflow.

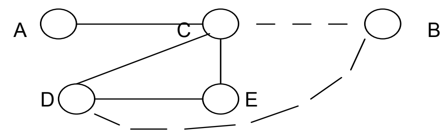
2. Shared Networks

Companies X and Y have an agreement to share their networks, also companies Y and Z have the same agreement, but X and Z have not. A call from Y1 to Y2 in Y may be routed through the X network if the direct link is not working. But a call to Y2 originated in Z must not be routed through X-nodes because X and Z have not agreed to share their network resources



3. Avoidance of Loops

Call from node A to node B (dashed lines are connections temporarily out of work)



A call from A to B must not be rerouted in node D via C because this can only end in a network loop.

A call originated in E to B can be rerouted in D via C because it is not yet known if a connection between C and B works.

All 3 examples show that the combination (source node, destination node) determines different routes.

Call Processing calls the DBAR mentioned above by supplying the input parameters source node number and destination node number (virtual nodes). The DBAR returns 2 LCR access right powersets. One access right powerset is used for removing access rights from LCR access rights and the other for adding specified rights.

Using the modified LCR access rights as usual enables choosing one route and prevent choosing another route depending on source/destination information.

The 3 level node number has the following structure:

DOMAIN	SUBDOMAIN	NODE

For example, the node number 1-2-45 has the number 45 in the node level and is member of sub domain 2 and domain 1.

A 3 level node number can be fully qualified (1-2-45) or partial qualified (1-0-0). In a partial qualified node number "0" is to be interpreted as "don't care".

It is possible to administer LCR access rights groups for groups of nodes. A pair of access rights is stored for source nodes (1-1-0, 2-0-0), that is all source nodes in sub domain 1-1 and all destination nodes in domain 2.

A general rule for a group of nodes is overruled by a more specialized rule, f.e. (1-1-7,2-2-4) with other LCR access rights.

The AMO KNLCR administers LCR access rights for source dependent routing.

The feature is independent of the numbering plan.

8.8 Expensive Route Tone

Within HiPath 4000 systems, the "Expensive Route Tone" feature is available for networked HiPath 4000 systems with LCR. This feature can be set up individually per station. The expensive route tone warns users if a normally cheap route cannot be switched, e.g. due to congestion in busy hours, and the call has to be switched via a more expensive route element instead.

- The "Expensive Route Tone" feature is available for users of all HiPath 4000 voice terminals
- The alerting tone is always applied by the user's home switch, i.e. if a breakout to a more expensive route element occurs in a remote transit node, an expensive route warning signal is transmitted back to the originating node (caller's node).
- The "Expensive Route Tone" feature can only be used with the following network protocols:
  CorNet NQ or DPNSS1 (CDG).

Once the caller's PABX system recognises that the required connection can only be set up via an expensive route element, an "Expensive Route" tone is sent to the caller's telephone. Telephones with displays will additionally show the message "EXPENSIVE ROUTE". While this signal is being applied, connection setup is temporarily suspended. This is done to allow users to decide whether to cancel their call or continue. If the caller accepts the more expensive route and continues dialling, connection setup is immediately resumed, and the expensive route warning is stopped. If the caller ignores the expensive route warning, connection setup is resumed after timeout, i.e. the system assumes that the expensive route has been accepted. The "Expensive Route Tone" feature is also applied for automatically dialled calls, e.g. via name key.

- The "Expensive Route Tone" feature is not applied for "implicit" calls, e.g. callback calls.

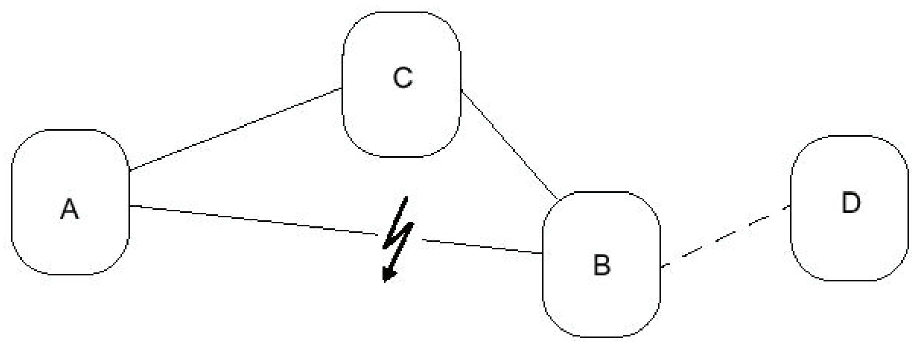
8.9 Dynamic Trunk Reservation

For each trunk group in a node an additional parameter for trunk reservation is introduced. Depending on the number of configured trunks per trunk group, the trunk reservation level k is computed. Alternate routing via a trunk group is only possible if at least k trunks are available.
Each trunk group of a route is marked if it goes directly to the destination node or via an alternate route. A call is treated as direct routed until an alternate route is used for the first time. From that moment the call is marked as overflow routing (even if a direct trunk group is used again).

Dynamic trunk reservation according to ITU T E.41 requirements

The function "LCR" provides with the new Traffic Dependent Dynamic Routing a mechanism, which can prevent a network from coming to overload situations.

Figure:Simple network configuration with the risk of an overload situation



The trunk group AB is completely blocked (out of service or all trunks busy).

The Figure above shows a situation, where the rerouted traffic from node A to node B via node C can possibly cut down the direct traffic from node A to node C. The solution is to reserve a number of trunks of the trunk group AC for the direct traffic exclusively. The reserved number of trunks depends on the number of configured trunks per trunk group and is calculated according to the recommendation E.412 of the ITU-T.

Note: In case of the direction AD, the part AB is also understood as a direct route because node B is only the transit node. This is to be considered on the configuration of the network and is to be described in the AMO Examples of Use.

Impact to the System or Network Provider
A homogeneous network in variant HiPath 4000 V1.0 or higher is mandatory.

8.10 LCR, Numbering Plans and Dial Plans (US Specific)

HiPath 4000 supports 3 numbering plans:

– 2 explicit 3-level OSI-compatible explicit numbering plans: E.164 and Private Numbering Plan

– an implicit numbering plan

The new numbering plan features include

– virtual nodes

– skip digits/overlapping numbers

– PNP without escape digits and prefixes

Virtual Nodes

Virtual Nodes are groups of subscribers with the same numbering plan data. The numbering plan data of the explicit numbering plans comprise of level 0 (L0), level 1 (L1), level 2 (L2) codes.

1. Location codes (US term) are the equivalent to HiPath 4000's level 1 (i.e., Type of Number = National) codes. For HiPath 4000, US public and private numbers level numbers (i.e., Node Numbers) equate as follows:
   - · L2 = country code (International number)
   - · L1 = area code (National number)
   - · L0 = office code + ext. # (local number)
2. HiPath 4000 supports up to 200 own virtual nodes per switch.
3. Numbers be capable of overlapping the ext. # most significant digits with the office code least significant digits. A Skip digits field for each L0 indicates the number of digits of the office code to delete.
   The example table below illustrates a logical representation of a system table that stores multiple Virtual nodes.
   Example Node Number Table (PNP based on a Public Number)

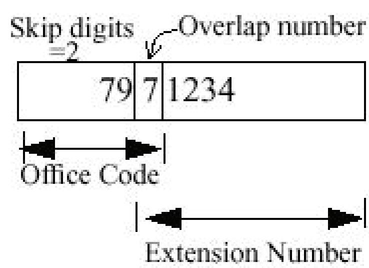
VirtualNode ID	L2 code (International / PNP Level2)	L1 Code (national/ PNP level1)	L0 Code (Susbcriber/ Local) (Office)	Skip Digits	
1-1-100	49	89	722	3	Munich
1-1-200	1	561	997	2	Boca
1-1-300	1	561	995	1	Boca
1-1-400	41	1	495	3	Zurich

4. Each subscriber, attendant, night station be assigned to only one Virtual Node. E.g., subscriber 71234 is assigned to virtual node 1-1-200 with L0 = 997, L1 = 561, L2 = 1 (entry 2 in the above table). The fully qualified private network number, when all 3 levels are administered, is 1-561-997-1234 representing country code (L2) + area code (L1) +[office code + extension number (L0)].
5. Virtual Nodes be able to span the network. That is, the same Virtual Node may reside on more than one switch, providing the same numbering plan data on different switches (exception: not the extension number, it must be unique).
6. The customer optionally may choose not to use country code, area code or office code as part of the private network number. The valid options are L0 only (with or without office code), L0 and L1 only, or L0, L1 and L2.

7. Fully qualified (i.e., only those levels administered) information numbers (i.e., calling, connected, redirecting and redirection) be transported over CorNet by default, different options can be administered.

Skip Digits and Overlapping Numbers

1. Any part, starting with the most significant number down to the least significant number, of the up to 6-digit extension number may overlap with the L0 number, but the overlappin part must not exceed the length of the L0 numbers. A typical example is a 7-digit PNP comprised of a 3-digit location (office) code and a 5-digit extension number where the most significant digit of the extension number is the same value as the least significant digit of the location code (e.g., 797-1234 where the extension number is 71234).
2. The number of digits to delete or skip (i.e., beginning from the leftmost digit of the office code) be administrable. The network administrator may assign the number of skip digits using a parameter while defining the a new virtual node. The skip digit value is the number of digits to delete whose remainder is the overlapping extension number. For example, given a 7-digit PNP whose most significant digit of the extension number overlaps with the least significant digit of the location code results in a offset of 2 digits. The diagram below reflects an L0 office code=797, ext. #=71234, the digit 7 is the overlapping number and the digits deleted are 79 therefore, the number of skip digits is = 2.



3. Skip digits be stored within the virtual node table. Each L0 table entry can have a different skip digit value. Skip digits must be the same value networkwide for a particular L0 (e.g., to ensure proper displays). Internally, the number of subscribers in this switch assigned to each vritual node is stored in this same table (i.e., population). Population is incremented and decremented with each subscriber that is added or deleted (respectively) for each of the up to 100 virtual nodes.
4. The deletion of the skip digits occur on the called number prior to entering digit analysis.
5. Skip digits be usable during the optimized display logic. For example, A, whose fully qualified number = 561-997-5555, calls B in the same switch with the fully qualified number of B; i.e., A dials "561-997-1234". The calling party number display on B's device shows just the extension number, e.g., "71234", and not the area code and office code.

QUANTITIES "Skip Digits" value in may range from 0 to the maximum length of the office code value. The default is "skip all", i.e. no overlapping. Skip digit values are used locally only and not sent across CorNet.

PNP without escape digits and prefixes

It is possible to use PNP numbering plan without escape digits and prefixes. This results in implicit numbers, that look like explicit numbers. With the above example virtual node table, this results in displayed numbers like 4989722 xxxxx, in comparison to "normal" PNP 9-11-4989722 xxxxx. 9 is the escape digit, 11 the L2 prefix.

8.11 Alternate Routing on Error

The feature "*Alternate Routing on Error*" is used in those cases, when a subscriber or personal attendant is not seizable, (not being in READY state). With this feature it is possible to call an alternate destination, which is assigned before.

Two categories of "*Alternate Routing on Error*" can be distinguished:

– AP emergency mode.

– Subscriber lines are in the state "Out of Service" (OOS).

The alternate routing mechanisms provide alternate destinations as alternate routing numbers in both categories.

– The first category of alternate routing numbers will be used exclusively for AP emergency.

– The second category will be used for all other OOS situations.

For both categories of alternate routing, it is possible, to use subscriber individual alternate routing numbers or source group related alternate routing numbers.

For called subscriber at checking time, one of the following 4 OOS situations can be valid or even a variable set of combinations. The OOS states, that are handled are as follows:

– An AP emergency condition exists between related APs of calling device and called subscriber, if connected over the IP-network.

– Dependability verifies unavailability of subscriber (device/ board error, stand-by switchover etc.)

– Subscriber/ board is switched off by AMO with the additional request for alternate routing.

– Subscriber (special flagged HFA phone) logoff from HiPath 4000 and logon at HiPath 3000 because of LAN/WAN error.

In case of an unavailability of the called subscriber, the alternate routing mechanism use the following realization principles:

– Validation of blocking situation at side of called subscriber
– Check, whether a request for alternate routing is necessary and possible
– Transport the request for alternate routing to calling device
– Evaluate alternate routing number at side of calling device
– Perform alternate routing from calling device

Alternate destinations in OOS scenarios can be either

– an alternate route via PSTN to former called subscriber number,
– a mobile number, to reach the subscriber in this way,
– alternate phone numbers or other foreign extensions.

Remarks and Restrictions

For both categories of alternate routing, the subscriber individual alternate routing number may be different to the entry in the source group related alternate routing table or not.

With respect to the feature ì*Autoset Relocate*î, if subscriber moves into another AP, the alternate routing number may not be usable anymore for that purpose, if it is designed as an individual complete number. So it may be necessary to modify this number additionally. In case of a source group related administration, no update is necessary.

When administrating the alternate routing number, only a few off_line checks can be done by the AMO. Therefore the Call Processing performs a plausibility check in order to avoid loops, before using this number for alternate routing. In case of diverse implausibilities the number will not be accepted.

9 Serviceability

9.1 System Administration

9.1.1 Access System for HiPath 4000

9.1.1.1 Access protection from HiPath 4000

With HiPath 4000 the PABX has improved protection against unauthorized access. Sessions can only be opened with the right user IDs. The user must now identify him/herself by his/her name and password making the corresponding user ID known to the system. Certain classmarks are assigned to each user ID allowing the user to execute certain commands. The ID number (user ID = UID) used to call up the commands is entered in the logbook. A priviledged user, the administrator, administrates the user IDs.

9.1.1.2 Advantages of the extended access protection

- user-specific COS assignment.
- **Monitoring**: it is known who performed which administration operations (logbook)
- All passwords are stored in **encrypted form** only (one-way function)
- The user can now **alter** his **own** password**.**
- The passwords are **secret** (unknown even to the administrator) and can be used network wide.
- AMO tasks of the **lowest class** (class 0) are now password protected .

9.1.1.3 Access system particulars

- Installation of new user IDs

  New user IDs (maximum number = 40) are configured using USER AMO. The user ID is automatically assigned by the AMO.
- Definition of user names

A user ID is defined by the user name (also called login name) entered on the user interface. A maximum of eight characters (capital letters, digits and special characters in any combination, with the exception of quotation marks ["]) are allowed. The user name must be unique in that it exists only once. The user then uses this name to log into the system.

The system administrator assigns, and if necessary, changes the user names in agreement with the users. Every user should have a unique user ID. In case of an unauthorized access the user ID used for this access can then be ascertained.

- Definition of preliminary passwords

  Newly configured user IDs are assigned a preliminary password by the system administration (exception: user without option P, see Options). A maximum of eight characters (capital letters, digits and special characters in any combination with the exception of quotation marks ["]) are allowed. During the first login the user is forced to change this preliminary password (exception: users without options C or F). The passwords are stored in encrypted form. They are, however, not checked for the minimum length, special characters used, combinations of characters and special characters, etc.! Decryption is not possible (one-way encryption). The passwords are not stored in plain text!

- COS assignment

  The system administrator assigns one or more COS to every user ID. These COS allow the user to execute certain commands (AMO tasks).

  A fixed "required COS" (class) is assigned to every AMO task. A user can only call an AMO task, if the class of this AMO task is included in his/her COS. If the COS is insufficient, an error message is displayed and the AMO task is not performed. The task can only be performed using a different user ID with this COS.

  The AMO task classes are permanently set in HiPath 4000. They are based on the hierarchical structure of the previous password classes. It is useful when assigning COS to comply with the passwort classes of PASSW AMO (see figure 1). The lowest COS is 0.

previous class	USER COS
5	0&&5
4	0&&4
3	0&&3
2	0&&2
1	0&1
0	0

Table 5 COS assignment with regard to password classes

- Definition of options

One or more options can be set for one user ID using the ADDITION parameter. The following options are possible:

L	Local use. The user ID can be used locally (CON1 to CON6 devices)
R	Remote use. The user ID can be used by the remote administration and maintenance (FAS device) (modem or LAN)
C	Change own password. The user can alter his/her password.
F	Forced password change. The user is forced to change a preliminary password.
P	Password required. The user has a password (without password means: login only with USER name without password prompting).
E	Erasable. The user ID can be deleted. (This flag provides the administrator with a protection against unintentional deletion, since only the administrator can set/reset or delete this flag).
T	Timeout. Forced logoff after timeout (after ca. 2 minutes)

During configuration of new user IDs the options L to E are set as default (i.e. all options with the exception of option T).

- Limitation of unauthorized login attempts

  There is a counter for every device and every user ID which counts the number of unsuccessful login attempts using a certain user ID. With each successful login attempt the counter is decreased. Every counter has a threshold value and a disable time. When the threshold value is exceeded the device or user ID is blocked for the configured disable time. The threshold values and disable times can be set differently for the IDs and devices.

- User forgets password

  If a user forgets his/her password, the system administrator can assign a new preliminary password. If the system administrator should forget his/her password, there is the following solution. Access to the ROOT ID is possible without password, provided the service dongle is plugged in and the corresponding code word was entered using CODEW AMO. Access to the LROOT ID is also possible without a password. In this case, however, the service dongle must have a special classmark.

IMPORTANT: Entering the service code word requires that at least one user ID (even with lesser than administrator rights) is still valid.

- Change password

  Every user can change his/her password using the PASSW function.

9.1.2 Remote Administration and Maintenance

- Remote maintenance includes maintenance functions offered by the service terminal. They can also be carried out from a common service center for several systems.
- Remote administration includes administration functions offered by the service terminal. They can also be carried out from a common service center for several systems.
- For access protection.

9.1.2.1 HiPath Teleservice

The HiPath Teleservice (HTS) supports safeguarding and operational maintenance of the system from a regional service center. HiPath 4000 is accessed from a HTS station via the public telephone network. The data link from the service center to the HiPath system and vice versa is established by means of automatic dialling. ID or password replacement and a callback function ensure that no impermissible connections are set up. In addition, remote access to HiPath 4000 systems can be blocked by the customer.

Functions of the HiPath Teleservice

The basic HTS functions offer the same capabilities to both service technicians in the service center and service technicians in the field. They essentially consist of the following:

- Automatic transfer of detected faults.
- Ability to call up all operational commands implemented in the HiPath 4000 system for administration and maintenance purposes.

Access of HiPath Teleservice (HTS) to administration and maintenance

The access procedure of the HTS to the administration and maintenance is as known and is controlled via the access table. This table contains all AMOs and tasks barred for HTS access. Customers can change and interrogate this table from the operating terminal.

The access table is administrated by AMO. This AMO also allows a general switch ON or switch OFF of the HTS AMO start. In addition to the access table this interface is also protected by call back procedure.

Automatic transfer of error messages

Faults detected and analyzed by the HiPath safeguarding system are transferred to the service center by means of AFRs (automatic fault reports). The actions then taken serve the following purposes:

- Elimination of the fault from the service center (e.g. patches).

- Deployment of a service technician to eliminate the fault on-site.
- Logging of fault statistics.

With HiPath these error messages can be transmitted to 3 different destinations at once.

The following destinations can be configured:

- HTS
- HiPath 4000 Manager
- SICON (signalling control)

It is also possible to configure more than one destination of the same type (e.g. 2 x HTS, 1 x HiPath 4000 Manager).

Troubleshooting

The reported faults are buffered by the automatic fault report (AFR) in a fault data file on the ADS hard disk. The individual fault entries are provided with priority codes, making it possible to:

- Immediately report high-priority faults to the service center.
- Collect lower-priority faults according to time or quantity and then report them to the service center.
- Self-correcting faults stored on the HD (e.g. line alarms) are deleted by an OK message.
- Multiple transmission of identical faults is prevented.

However, for systems still in the field trial phase, all error messages are required in order to localise and resolve errors as rapidly as possible.

For this reason,

1. all error messages determined by the dependability system to be sporadic errors can be stored in the History file on the hard disk,
2. all other error messages can be output to the service terminal's output device (corresponds to the existing solution applied to the dependability alarm concept).

   In order to ensure that errors/faults are reported as rapidly as possible, the following output settings can be made for the automatic fault report system via the HISTO AMO:

   Output of all error messages / alarm messages

   – Output of all error messages / alarm messages with a specific error message / alarm message number

  - Output of all error messages / alarm messages for a specific station number / dialled code
  - Output of all error messages / alarm messages in a specific alarm class
  - Output of all error messages / alarm messages for a specific port equipment number (LTG/LTU/board/cct.)
  - Output of all error messages / alarm messages with a specific priority
  - Output of all error messages / alarm messages with a specific functionality
  - Output of all error messages / alarm messages with a specific call code / exception code and a specific reader address

9.1.2.2 Applications of the HiPath Teleservice

- Faults reported over the telephone
  - If possible, fault elimination by remote access via HTS
  - Rapid error elimination
- Automatically reported faults/alarms
  - Fast error detection
  - If possible, fault elimination by remote access via HTS
  - Preventive maintenance
- Remote administration

  Fast response to changes requested by the customer
- Remote software correction

  Supply of systems with patches
- Problem systems/startup
  - Monitoring
  - No personnel tie-up on site
  - Service technician support on site

Data protection is supported by the following actions:

- Blocking of changes to any DB data (e.g. personal data). The data to be given this special treatment can be determined in the HiPath system by means of an AMO.

- For control purposes, it is possible to log all remote access operations to HiPath 4000 from the service center on a local printer (at the HiPath system).
- Callback function for preventing impermissible access to the HiPath system

9.1.2.3 Automatic Collection Of Node Numbers

'Automatic Collection Of Node Numbers' is a serviceability feature.

With the feature 'Automatic Collection of Node Numbers' the node information of all adjacent HiPath 4000 nodes, physically connected to a specific HiPath 4000 node, is available, and it is possible to request this information using the new AMO KNTOP. The node information contains the 'Physical Node Number' and the 'Physical Node Access Code' and is related to the trunk / B-Channel group. The node information is built if a trunk is brought into service and also periodically during the trunk is in service. The feature uses the Layer-3 protocol CorNet NQ for inter node message exchange. That means, the feature needs inter node user traffic (call setup / connect) in order to build up the adjacent node information tables.

The feature is only active, if the central system data NWTOPTIM is configured to a value different from 0 (AMO ZAND).

Feature Description

With the help of HTS-SMART, Service is able to visualize the customer network topology in the service center, in order to improve analysis of faults across nodes. The adjacent node information needed to construct a network topology in HTS-SMART is determined automatically in each HiPath4000 node if the feature is enabled and is accessible at any time.

The solution offers the following options:

– All the information on the HiPath4000 adjacent nodes is collected in each HiPath 4000 node.

– The information is collected without manual activity, i.e. is programmed.

– The information contains all data important for the network topology (physical node numbers, physical node access codes, locations, etc.)

– The partner HiPath 4000 systems are under periodic monitoring by a program, and the information is automatically updated in the case of failure (broken line, partner power off etc.).

– After each line fault has been cleared, the information is collected again by the periodic partner monitoring.

– The applications (HTS-SMART, etc.) have access to the information via AMO KNTOP. By successively querying all nodes, the application can determine the network topology.

- The time until a topology change is actualized in the adjacent node information table depends on the number of trunks, the traffic load and the system polling timer (NWTOPTIM in AMO ZAND). HTS-Smart requests the information in approx. 10 min. time periods.
- Pedecessor systems of HiPath4000 do not support the automatism. These node informations can manually be added to the adjacent node information table (by AMO KNTOP)

Environment/Restrictions

The feature (automatism) can only be used in homogeneous HiPath 4000 networks. The solution described is viable in the following environment:

- The feature functions completely only in a homogeneous HiPath 4000 network consisting of only HiPath 4000 nodes .
- A heterogeneous network can be depicted in the topology if the table with the adjacent node information is administered manually by AMO KNTOP (in cases where the adjacent node is not a HiPath 4000 node).
- The HiPath 4000 nodes have to be connected via CorNet NQ with segmentation 8. If a different protocol is used, the statement referring to the heterogeneous network is valid.
- There must be traffic on the network.

The network may contain multiplexers and other backboning systems (e.g. ATM), as long as they transfer the Cornet-NQ protocol transparently. Multiplexers are not visible for the HiPath 4000 and therefore not shown in the topology but the nodes behind the multiplexer / backbone system are recorded and appear in the topology.

9.1.3 AMO Logbook

The "logbook" feature records all the administration and maintenance operations performed in the system over a definable period of time. This allows maintenance personnel to research failure causes and reconstruct previous command sequences.

- The log book data is stored in a restart-safe and reload-safe database file system on the HiPath 4000 hard disk. The log book information can only be read by authorised personnel, with the aid of the LOGBK AMO.
- Despite being protected against deletion, the data is always deleted when the hard disk is formatted. This complies with the German data protection laws.
- In system networks, the logbook data is always stored on the hard disk of the PABX in which the feature is activated.

9.2 Serviceability Platform Unixware 7

9.2.1 Single Point of Access

Features

The main features of the new service platform are as follows:

Open Application Platform at the ADP and Open Service Interfaces

- Single Point of Access to HiPath 4000 system
- Asynchronous Protocol PPP for HiPath 4000 service
- Transfer and Network Protocol - TCP/ IP
- File Transfer FTP
- Web Protocol HTTP (e.g. for browser based administration)
- Fast remote/local switch access (V.24 asynchronous, Ethernet 10 / 100)
- Script Capability with (Restricted) Unix Shell Access
- Telnet
- Java

Fast Service Access

- Modem access up to 28,8 kbit/s
  – automatic bit rate adaptation
  – use of standard Modems
- LAN Access
- TAR Compatibility
- Transparent Data Transfer
- File Transfer Capability (Unix files, RMX files)
- Asynchronous Access (PPP)
- Capability to multiplex several sessions across single interconnection
- Internet access (planned)

Single Point of Access (SPoA)

Routing and firewall functions are integrated in UnixWare 7, therefore this service access can be used as the single point of access to remotely access corresponding internal servers and applications.

- SPoA means
  – single port
  – just one modem
  – single dial-up session
  – single log-in (planned)
- Access to interconnected server / applications
- Router function within the system
- Secure Access to UW7 basic functions
  – Four levels of access rights
  – Password Protection (dynamic)
- Firewall function within the system
- Suitable for all service tasks
- Call-back function (planned)

Web Based User Interface

- State- of- the-art integrated browser with service PC (HTS and HOT planned)
- Access to service functions via HTML web pages or via upload of a Java script
- In principle user PC sufficient
- All current software is available at the HiPath
- Easy to use (Intranet compatible)

Fast LAN Access at ADP (for on-site service, link to customer LAN)

A second LAN interface is available with the ADP dynamic packages to allow interconnection to a customer LAN. The usage of this access for interconnection of external servers is possible on a customer specific basis .

9.2.2 Software supply and Back-Up

Software Tool: delivery of HiPath software (RMX, UW7, Applications) from a central server with the Service PC (TAP) and/or on the HiPath 4000 UW7 SWS-Slice. There are two types of delivery of SW: delivery of new SW Revision Levels (RLC) or Patch Packages.

HiPath 4000 File Manager (HFM): the HiPath File Manager offers the possibility to copy, rename and delete files on the HiPath as well as to view text files. File systems referred to are RMX, UW7, TAP, server (reachable via FTP)

Back-Up: The back-up functionality enables parts of the HiPath SW (without applications, only RMX and UW7 with UW7 applications) to be saved in another location (e.g. on a server on the connected LAN) and retrieved.

Description:

The service function supports the following additional features in UW7-SP2 of HiPath 4000.

Features with SF-BACKUP

- Introduction of Gold Load Area (GLA) to RMX
  With the UW7-SP2 within HiPath 4000 the GLA has be used for automatic fallback by RMX.
- GUI for Gold Load Area
  The GUI of SF-Backup of SP2 is also display the currently active load area and medium of the switch and the creation date and time of the GLA.
- Introduction of a Basic HiPath 4000 System (Rescue SW)
  In case of no stable SW release is available on the HiPath 4000 Harddisk, neither the primary Load on the :PDS nor the gold load on the GLA, RMX has try to bring up an reduced switch which allowes an remote access by UNIX means. For this the startup the ADP part is sufficient including the unix loader and TCP/IP.
- Support of 'Logical Bakkup/Restore',
- Complete SF Backup' functionality. (e.g. multiple data backup sets on HD).
- Support for a Customer Backup Server (over FTP or NFS).

SF-SWT, SF-SWA and SF-HFM
With UW7-SP2 the following Service-Functions already available with UW7-SP1 has be enhanced:

Features with SF-SWT (UW7-SP2)

- Transfer of VKs and OPS
  Transfer of VKs (Vorabkurrekturen, which have been produced between two PPs) and OPs (Optional Patches) has not be supported by the SF-SWT. Activation of VKs and OPs has not be supported by the SF-SWT or SF-SWA. Consecutive Transfer and Activation of HiPath 4000-SW on Different HiPath 4000-Switches
- Consecutive transfer and activation of HiPath 4000-SW (PPs and RLCs/SAs) on different HiPath 4000-Switches has not be supported by the SF-SWT itself.
- Transfer of HiPath 4000-SW to TAP without Connection to HiPath 4000
  With UW7-SP2 the transfer of HiPath 4000-SW (PPs, RLCs/SAs) from the SWS-Server to the TAP without connection to a HiPath 4000 is possible.

- Enhanced Log-File
  With UW7-SP2 a selection of log information has be introduced. Also a complete information about an error situation has be provided.
- HiPath 4000-SW Server-Server Transfer
  With SF-SWT it be possible to transfer PPs/ RLCs/SAs from one server to another server like ITSC-SWS-Server to the REGION-SWS-Server.

Features with SF-SWA (UW7-SP2)

- Activation of VKs and OPs
  Activation of VKs and OPs has not be supported by the SF-SWA. The activation has to be done using the appropriate AMOs.

Activation of SW for Servers (Applications)

9.2.3 HiPath Inventory Management

The HIM (HiPath Inventory Management) feature provides complete information about the HiPath hardware and software configuration.
The HIM is completely integrated in CM (Configuration Management) and include all features from HIM Step1, Step1a and HIM Step2. Furthermore, these data are kept synchronous with the switch data via notify mechanism.

HIM Step 1 and 1a

- Information about System - Configuration (HW and SW) available within the System (interfaces to configuration tools)
- Retrieval of database about current HiPath 4000 HW/SW configuration data (system data)
- Enquiry to individual configuration data
- Cyclic update of configuration database, configurable schedule
- Integration of projection information into database
- AMO-REGEN

9.2.3.1 The following features are removed from HIM

- AMO Execution
- Extra Functions/Input Interface for IP Configuration
- DB Update Time
- SNS Products

9.2.3.2 Features

– switch data
– special switch data
– Configuration of Switching Unit (SWU) Periphery
– Configuration of Administration Data Processor (ADP) Periphery
– Central Switch Data
– General Switch Data
– Features
– APS and patches
– SW Version
– APS information of service functions

9.2.3.3 Restrictions

The following requirements are not be realized:
An interface to retrieve the information on current board expansion for external application to calculate the total electrical power supply requirement.

No information about the pseudo boards e.g. CDG, PNE and PNG and the trading boards except SLMT are provided by HIM.

9.2.3.4 Information about Loaded SW and LW

The HiPath 4000 system software determines information about LW and SW currently loaded into the different boards. The information is stored into different hard disk files. They are read and displayed on the GUI by SF-HIM (Hicom Inventory Management).

The following information is available:

– Boot-FW for all DP-boards
– RMX-System-SW for all DP-boards
– Peripheral Board LW for all peripheral boards
– Central SIU Board LW for all central SIU boards
– Coprocessor-LW for DSCX boards

9.2.4 Dynamic Traffic Monitoring

Dynamic Traffic Monitoring (DTM) is a service function and is part of the Serviceability Package 2 (SP2) for the HiPath 4000 Manager. DTM is used for fast detection of bottlenecks in telephone networks and to measure traffic flows in the telephone network.

DTM provides a momentary status of the specified trunk groups or makes a monitoring job on them. The following trunk states are distinguished by DTM:

– free
– busy
– and other trunks.

The trunk is free if it is configured and is not occupied by a call. A free trunk can accept a new call.

The trunk is busy if it is currently allocated by a call. A busy trunk can not accept a new call.

The trunk is other if it is not available because of malfunction or wrong or missing configuration.

The status information of the specified trunk groups can be displayed at that very moment the user requires it (snapshot) or can be collected during a predefined time interval with a given frequency into a result database (monitoring). In both cases the results will be displayed on a web based interface.

Features:

Dynamic Traffic Monitoring is realized as a UinxWare 7 application. The application can be found under "Fault Management" à "Switch Diagnosis Support" à "Dynamic Traffic Monitoring". The application has a web based graphical user interface (GUI).

The application provides the following functions:

– Selection of the trunk groups
– Snapshot of the selected trunk groups
– Monitoring of the selected trunk groups
– Administration of the monitoring jobs
– Display of the monitoring jobs

Selection

The selection function provides an easy method for the user to mark some trunk groups for the snapshot function or for the monitoring function. The complete list of configured trunk groups is displayed with check-boxes for each trunk group.

The user can select up to four trunk groups for the monitoring function and unlimited number of trunk groups for the snapshot function. The trunk groups are displayed with the following informations:

- Check-box
  This control is used to mark individual trunk groups for snapshot or monitoring.
- Trunk group number
  The identification number of the trunk group.
- Trunk group name
  The text description of the trunk group.

The selection of the trunk groups can be done by selecting or deselecting the corresponding check-boxes one-by-one or with the help of the "Select all" and "Deselect all" buttons.

The "Snapshot" function, the "Monitoring" function or the "Update" of the trunk group list can be also started from the selection.

Snapshot

The snapshot function displays the actual traffic statistics of the previously selected trunk groups. The following values are displayed per trunk group:

- Trunk group number
- Trunk group name
- Configured channels
- Busy channels
- Other busy channels
- Free channels
- Maximum number of channels

The number of busy, other busy and free channels is also displayed as a percentage of the configured channels.

The "Snapshot" button refreshes the snapshot page by executing the snapshot function of DTM on the selected trunk groups again.

Monitoring

The monitoring function gives the possibility to specify various parameters for the monitoring job of the selected trunk groups. The monitoring job will be created finally with the specified parameters.

The following additional parameters can be specified for the monitoring job:

- Starting method
  Immediate or scheduled.

– Start/stop date and time
  Only for scheduled monitoring jobs.

– Interval
  Measurement frequency for the monitoring job (10, 20, 30 or 60 seconds).

– Memo
  Text description for the monitoring job.

Administration

The administration of the monitoring jobs displays the most important informations about all configured monitoring jobs. They are

– Job number

– Creation date and time

– Trunk groups

– Status
  Scheduled, running or so topped.

– Start/stop date and time

The administration of the monitoring jobs allows the user to execute the following functions:

– Job information
  Additionally displays the memo field of the job and the measurement interval of the job.

– Stop
  Only for running jobs.

– Delete
  Only for scheduled or stopped jobs.

– Display
  Only for running or scheduled jobs.

Display

The display function provides a graphical presentation of the measurement with the following functions:

– Fade in/out
  Displays or hides the curve of the selected trunk group.

– Scrolling and shifting
  Any parts of the measurement can be easily displayed with the help of the scrollbar. The selected part can be locked so that no scrolling will happen during new measurements.

– Tooltip
  The exact numerical values of the measured traffic will be displayed.

9.2.5 Local Alarm Agent

- HiPath 4000 LAA works as a local agent for the remote service system (HTS)
- IP ping to external server
- Schedule configurable
- Configuration of IP addresses
- Automatic generation of alarm when ping failed
- Forwarding of alarm to AFR facilities within HiPath 4000 (dependability) and, in consequence, to HTS
- First step in SPoA integration of external respective internal server and applications
- Browser based user interface

9.2.6 Hardware and Symptom Diagnosis (HSD)

HSD is a complex consisting of a set of two components: The HSD User Interfaces and the HSD server.

The HSD User Interface is representing a typical client / server architecture. This User Interface's client side interacts with a set of Common Gateway Interface (CGI) functions handling the network communication tasks between the browser and the HSD server process itself.

The User Interface can be split in the following functionality complexes:

– Hardware Diagnosis

– Symptom Diagnosis

Each complex is communicating with the server and the RMX.

9.2.6.1 Hardware Diagnosis

System State
display overwiew of processors states / alarm ...)

Alarm Summary Table
dispay all alarms of the system

functions and links Relating to Marked single Alarm

reset Alarm
reset the dependant alarm(if allowed) and refresh display

diagnosis
start HW-Diagnosis for related Alarm and display diagnosis report

all related PEN
display table of alarm releted units
(format and links of result table like Fault summary table)

defect related PEN
display table of alarm releted fault units
(format and links or result table like Fault summary table)

all fault messages
display table of alarm related error messeges (last 10 days /maximal 500)
(format and links of result message table)

fault messages of last 2 hours
display table of alarm related error messeges of last 2 hours (maximal 500)
(format and links of result message table)

Fault Summary Table:
displays all fault states of the sytem hardware units in a table

functions and links Relating to marked single Fault

activate
activate related unit and refresh display

deactivate
deactivate related unit and refresh display

restart
restart related unit and refresh display

diagnosis
start HW-Duagnosis for related fault unit and display diagnosis report

all fault messages
display table of fault unit related error messeges (last 10 days /maximal 500)

fault messages of last 2 hours
display table of fault unit related error messeges of last 2 hours (maximal 500)

Search FM (display of error messages)
Display screen to input filter parameter for error message search. After start searching, a table with short message text releted to the input parameter will be displayed. (max 500 of last 10 days)

Functions and links relating to marked error message

Display:
displays own screen with the full error message (original format)

Description
display the ALFE description of the related Error message

HW Statistics
This link enables the display of the table with statistic information for all hardware related messages

Functions and links relating to marked error message

Description
display the ALFE description of the related Error message

SW Statistics
This link enables the display of the table with statistic information for all software related messages

Functions and links relating to marked error message

Description
display the ALFE description of the related Error message

9.2.6.2 Symptom Diagnosis

For all problems an Input screen has to be filled to describe the problem. After start of the Siagnosis an Diagnosis report will be displayed with following parts.

– Symptom inputs
  main paramater to describe the problem (from input screen)

– Diagnosis description
  Result statement of the diagnosis.
  (found reason of problem in hardware or software configuration)

For defined symptoms only following info may be displayed additionally if no obvious reason for the problem is found.

- List of system unit faults related to the problem
  (format like Fault Summary Table)
- List of system error messages related to the problem
  (format like Message table)
- List of AMO commands which have been started the last days before and may have be in relation to the problem:
- Advisory info to related to the problem.

Following problem areas are defined:

- Phone Problem:
  Covers all symptoms for a single station. Input has to be mandatory to define the station (by number or PEN) and selected symptom.
- Voice Problem:
  Covers all symptoms for a intern connection beetween two (voice)stations. Input has to be mandatory to define the A-station (by number or PEN), define the B-station (by number or PEN) and selected symptom:
- Data Problem:
  Covers all symptoms for a connection beetween two (data) stations. Input has to be mandatory to define the A-station (by number or PEN) and selected symptom. B-station can be defined optional(by PEN or number)
- Attendant problems;
  Covers all symptoms for a attendent stations and features. Input has to be mandatory to define the station (by number or PEN) and selected symptom. B-station can be defined optional(by PEN or number)
- ACD problems;
  Covers symptoms for depending ACD features. Input has to be mandatory to define the station (by number or PEN) and selected symptom. B-station can be defined optional(by PEN or number)
- Callbridge problems;
  Covers symptoms depending Callbridge features. Input has to be mandatory to define the station (by number or PEN) and selected symptom.
- Phonemail problems;
  Covers symptoms depending Phonemail features. Input has to be mandatory only to define the selected symptom.
- CDR problems;
  Covers symptoms depending CDR features. Input has to be mandatory only to define the selected symptom
- Networking problem
  Covers all symptoms for a extern connection beetween two (voice)stations. nput has to be mandatory to define the A-station (by number or PEN) and selected symptom.
  The extern destination has to be defined by dialed number, or (if known) the dependant trunk device (by PEN or Trunkgroup number or Route number)

For all Problems the date and time the problem occurs may be defined optionally. This will be used for searchin releted System error messages.

9.2.7 Realtime Diagnosis System RDS

Realtime Diagnosis System provides telephony fault localization for station and data lines and limited trunk fault reporting capabilities for trunk facility problems of Hipath 4000 systems. It provides tools and features to help reduce the time of solving trunk problems, as well as enabling solving such problems more efficiently. These features significantly enhance remote diagnostic efficiency of telephony problems, and reduce the need for test equipment. RDS is accessible entirely by internet browsers. RDS provides the following features:

Monitoring trunk/line/data line
This feature allows viewing the call processing and device handler events as they occur. The user can watch what dialing and how the switch is reacting, which numbers are out pulsed etc. This feature allows the user to understand complex call scenarios online.

ISDN Trace
This feature applies to ISDN trunk interfaces and ISDN devices and allows the user to capture and display the decoded layer 3 messages as they occur. So the user is able to identify protocol mismatches very fast, because of easy accessability of the interpreted layer 3 ISDN Messages

On-Demand Facility Tests
These tests facilitate testing between the HiPath 4000 and the facility provider by providing standard transmission tests, such as sending and receiving test tones.

RDS provides the following test functions:

- Bit Error Rate Test and Networkwide BER-Test:
  The BER test performs a bit error rate test on a channel basis. This test is a problem isolation tool, not an initial span qualification measuring device and always runs at 64 Kbps (respectively in the US at 56 Kbps). The network-wired BER Test will be used for establishing a network wide connection through a HiPath 4000 Network with more than 2 switches. The feature works for digital trunks only. The connection will be established switch by switch.
- Echo Return Loss/Singing Return Loss (ERL/SRL) Test:
  The ERL/SRL Test measures the echo return loss and singing return loss of 4-wire to 2-wire transmission points..
- Send Signal Test Control:
  The Send Signal Test transmits test tones to other switches or remote service personnel. It runs in either forced or interactive mode.
- Receive Signal Test Control:
  The Receive Signal Test measures incoming test tones from other switches or remote service personnel.
- Loopback Test Control:
  The Loopback Test measures round-trip transmission loss over a 4-wire trunk circuit.

- Trunk Rolling:
  This feature is only available for specific boards. It enables you to exchange or "roll" trunk facilities on TMC16 boards.

9.2.8 Trafficmetering with CDRe

The Hipath 4000 feature "Call Detail Recording enhanced - Trafficmetering" allows customers (or Service) to obtain an overview of all the important processing functions carried out within a HiPath 4000 system or network over a given time. The main areas of application are:

– · Dimensioning the exchange, tie-traffic and server trunk groups
– · Determining the traffic carried statistics and the busy hours
– · Determining the frequency of busy or otherwise unsuccessful call setups
– · Determining the average call duration and hold times for incoming Trunkcalls to the attendant console
– · Determining the user acceptance of some HiPath 4000 features on the basis of user activity data

Features:

With Hipath 4000, the CDR (Call Detail Record) has been enhanced for more detailed call record output.

The enhanced CDR can be used for allocation of call costs to the appropiate paying party and also for measurement purposes of the call handling performance of the system.

In addition to the call charges, the CDR output contains information about call duration and intervals, source and destination numbers of the calls, call routings (LCR) etc., which can be transferred to an external billing system.

This ensures that each record contains enough information about every step of a connection (call situation) e.g. the connection is traced precisely in the system, the affected subscriber numbers, some affected active features and the correct ringing time for each call processing step are recorded.

All the information for a specific call situation is available in a set of CDR s corresponding to the specific call situation.

For example:
Party A has activated CF to Party B and Party B has activated CF to Party C.

With an incoming call to Party A, that is accepted by Party C, one group of call detail records associated with the call is identifiable with an ID and a CDR is produced for each of the separate connection legs.

This feature is inactive as default and is separately marketed to ensure data security/personal security .

Statistics data can be collected

1. for the following HiPAth 4000 elements:
   - Individual users (stations)
   - Hunting groups, with/without call queues
   - Individual attendant consoles
   - Attendant console groups
   - Individual exchange trunks
   - Individual tie-trunks
   - Exchange trunk groups
   - Tie-trunk groups
   - Server trunk groups
   - External connections
   - Internal connections
   - Tie-trunk connections
   - All connections
   - Incoming traffic only
   - Outgoing traffic only
   - Bothway traffic
   - Usage of some specific HiPath 4000 features
   - Any combination of the above
2. for the following connection setup phases:
   - Average waiting time for calls in call queue
   - On hold in case of incoming Trunkcalls to the attendant console
   - Average idle time per station
   - Average busy time per station (talking state)
   - Call forwarding
   - Call Pick-up
   - Conference

3. Total number of calls

   – Successful connection setups, unsuccessful connection setups and busy connection setups per time period

   – Distribution of ringing times (duration of ringing) between successful and unsuccessful connection setups (in intervals of <5 secs. to >60 secs.)

   – Average talking duration, ringing duration

   – Average waiting time per time period for incoming Trunkconnections to the attendant console

   – Minimum, maximum and average duration of talking, ringing or waiting time per time period

   – Distribution of call states (idle, talking, ringing)

   – Traffic volume over specific time periods (busy hours)

   – Overview of HiPath 4000 feature acceptance for some features (user activity)

   – Incoming call statistics per attendant console or attendant console group, including distribution among exchange lines, internal lines and personal lines

   – SWU control processor load

4. AC Activities

   – Switching to night service

   – Pulling the jack

   – Pluging the jack

   – Abandoned calls to the attendant

CDR collects the call data and generates a call data record for every stage of the call when the destinaton changes. In addition, this feature can be used to limit the generation of data records to certain objects.

Retrieval of call details via file transfer from DMS-COL

All the recorded call data records of a call with each other at different points of a network are considered. The call data records of a call which are recorded at different nodes of a network contain data, to run an analysis to ensure that this call can be clearly traced in the network. The call ID is for identifying a call via multiple network nodes.

Restrictions:

– Data is not recorded by CDR until a call segment has been completed, a limit value reached or the call terminated. This means it is not possible to use CDR to observe ongoing calls, for it does not provide information on calls until AFTER the calls have been ended. Consequently CDR cannot be used for supplying online statistics, i.e., statistics to be used for deriving statements on the current state of the system.

– Statements can only be derived from CDR's data records retrospectively.
– Data records for the traffic measurement are only recorded if the call's active user (CALLIPTY) or passive user (DESTPTY) is included in the list of objects.
– Connection setup time will not be recorded
– Connection release time will not be recorded
– Waiting time for parked calls will not be recorded
– Availability of metered objects (up-time) can not be determined with this feature
– The callstates during connection setup and

9.3 Controlled Line and Trunk Select

For maintenance service purposes, this feature facilitates the controlled select of analog/ digital lines for CO lines and tie lines, as well as music on hold and announcement equipment. It enables service enhancements on commissioning and trouble shooting connection paths.

CLTS makes it easier to administer the network as the functionality of analog and digital lines, CO lines and tie lines can be selectively tested .

Analog voice mail connections and digital trunk circuits that belong to nailed connections are excluded.

CLTS can be used across the HiPath 4000 network. The seizure of lines or subscriber line circuits by selecting the port equipment number is only possible from CorNet-NQ networks.

Activation of the Feature

The selection of analog and CAS trunk circiuts (as a CO line or tie line) by B-channels on digital lines and by SLMA and TMOM ports is only performed by authorised users by entering (dialling) the port equipment number and, if necessary, the B-channel in a defined format.

With CLTS the selection of TMOM ports is only possible for connected announcement.

In the transit case, the dialling of an equipment port number can only be continued if the previously selected circuit was a CorNet NQ circuit.

If, in the CorNet NQ transit case, the outgoing circuit to be selected is dialled by means of a phone number and not via the equipment port number, equipment port number dialling cannot be recommended in the next system.

CLTS has to begin with selection of a port equipment number.

Affected Terminal Units

CLTS has been made available for use on Optiset-E and optiPoint 500 terminal units.

It is also possible to use analogue terminal units (DTMF-capable or with external DTMF sender).

CLTS is also supported by the attendant console.

Functional terminal units DSS1-EG and CMI-EG are excluded.

Serviceability and Authorisation

As not only the granting of authorisation but also the usage of CLTS is a service task, each usage is logged, i.e. a log message (maintenance message with plain text) is output on each usage.

This message contains the phone number of the user and the equipment port number dialled in the originating system. The message also contains for each transit system HiPath 4000 that has been reached with CorNet NQ, the incoming equipment port number seized and the equipment port number or phone number with which the incoming line that has been seized was reached.

If the user has identified himself/herself across the network by means of a PIN, the PIN-HOME number is output in the log message in addition to the phone number of the line used.

All messages also indicate the usage of CLTS.

Dependency on HiPath 4000 Versions in the Network

The system in which the user with CLTS authorisation is administered, the system to which the authorised user, if necessary on another terminal unit, identifies himself, and all other systems in between must be operated with HiPath 4000 .

9.4 Flag Trace

Flag Trace improves the serviceability of the HiPath and shortens response times on connection malfunctions. Diagnostics and troubleshooting in service and development are thus improved.

With the aid of **Flag Trace** , selected (with a malfunction or correctly functioning) connections can be traced in the individual systems in a HiPath 4000 network.

This way it is possible to determine which tasks and which paths are used in the network.

The trace data is call data that is recorded and analysed for performing diagnostics on faulty connections.

The following features are impacted :

Callback, call forwarding (CFU / CFNR / CFB), camp on/override hold, intercept, transfer/pick-up, call forwarding - after transfer , hold - toggle, conference, redial, call back, transfer, call queue on the attendant console, hunt group call queue and team-network-wide.

This is a network-wide feature and requires networking using the CorNet NQ protocol, all other signalling protocols cannot signal the information necessary.

The network-wide trace stop makes it possible to deactivate the standard tracer to another system from a node's operating terminal.

9.4.1 Activating and Deactivating the Flag Trace

Activating Flag Trace

The flag trace feature can be activated in two ways:

- A station (all system terminal units, functional terminal units and trunks) can activate a flag trace by prefixing a code, i.e. during call processing, all messages to the trace points that are related to call setup initiated by the station in all nodes in the HiPath 4000 network are recorded by the tracer, and are available for further evaluation in the usual way in the nodes over which the call is connected (prefixing a code for CO-trunks not possible due to security reasons)
- A terminal unit/trunk (all system terminal units and trunks) can be marked using the AMO (Administration and Maintenance Order) Trace. In the case of digital circuits, marking can be performed by B-channel group (line). Each incoming/outgoing call that starts from this terminal unit or that seizes the terminal unit with an incoming call, is then treated as if the flag trace code has been dialled by the user. The trace data from the traced call are then available in the local trace buffers in the nodes involved in the call, and can be read from there using the appropriate trace commands as previously.

In the case of network-wide calls, the node numbers of the nodes involved in the call are provided on the operating terminal of the node to which the flag trace initiator is attached.

The settings for the flag trace (message length, etc.) are locally administered in exactly the same way as for the standard tracer.

Deactivating Flag Trace

The flag trace is active by default on all systems. This feature can be deactivated, if required, using the Control AMO (Administration and Maintenance Order).

It is not necessary for the user or any other event to deactivate the flag trace if the prefix is dialled.

On terminal units marked using the AMO, the flag trace is automatically reactivated when the terminal unit is used again. It is not necessary to mark the terminal unit again using the AMO. If the flag trace is no longer required on a terminal unit, the marking of the terminal unit can be withdrawn using the Trace AMO.

9.4.2 Flag Trace and the Handling of Call Processing Equipment

The following call processing equipment is handled by the flag trace feature:

- Analog terminals
- Digite subunits
- HiPath 4000 Pro Center
- Parking console
- Attendant voice subunit
- Voice mail
- Special equipment
- Analogue circuits
- ISDN circuits
- Analogue networking circuits
- S0-Bus (functional terminal units)
- CLC connections

9.4.3 Flag Trace Watchdog

Flag trace watchdog now allows to examine the HISTA file of the HiPath 4000 network node from which the activation of the current flag trace originated. For stored in this file are the acknowledgments of all the nodes (terminal and transit nodes) which a connection passes through while it is being set up. The node numbers contained therein are extracted and transferred to the user.Operation is via a Web browser.

Now it is possible to define selections. This reduces the volume of data to be transferred and speeds up the process (the enormous flood of messages can cause the RMX trace files to grow to a size of several megabytes within a matter of seconds).

Now the transfer can begin in HTS Hipath 4000. On the HTS client an applet is started which sets up PPP or LAN connections to all network nodes involved, makes the local trace files available and then transfers them to the client.

9.5 Alarm Concept

Reduction of System Alarms

This feature consists of two sub-functions:

In case of LTU failure secondary alarms will be avoided automatically from now on. With the snapshot function the Service engineers have the possibility to pare the current alarm situation down to what are genuine alarms at any time.

AM orders would ensure that all SWU peripheral units that had the status NPR (Not PResent) at the time the snapshot was taken, would be excluded from alarm monitoring. This measure eliminates SWU alarms that resulted from missing boards, terminals or trunk connections.

The remaining alarms are only the genuine alarms caused by real error situations. The units excluded from alarm monitoring at the time of the snapshot would be included in alarm monitoring again if:

- the snapshot function were deactivated. This would then apply for all units which were excluded form alarm monitoring because of snapshot.
- a unit were put into operation automatically (e.g. if a faulty-line alarm were terminated) or manually by a Service engineer (e.g. if a board or terminal were inserted).

Avoid secondary alarms in case of an LTU failure

If an LTU fails, only the specified alarm "LTU FAILURE" will be triggered. The secondary alarms will be avoided automatically with the exception of some alarms with system wide impact (e.g.: ALUM alarms).

Reduction of System Alarms with Snapshot

With this Function, it is possible to start the snapshot function at any time by means of an AM order. The snapshot function can be activated repeatedly. This means that the delta set of units that are in the NPR status is always excluded from alarm monitoring.

There are two ways of putting units (that have been excluded from alarm monitoring by a snap-shot function) - back into alarm administration.

1. Automatic reinclusion

   Those units that have the status NPR go into operation (by being inserted/ connected or having a faulty line alarm terminated) and now have the status READY or any status other than NPR.

2. Manual reinclusion by means of an AM order

   Deactivating the snapshot function would have the effect of reincluding all those units, that had previously been excluded from alarm monitoring by the snapshot function.

User Interface

Users can activate or deactivate the feature (snapshot function) via AMO VADSU. In other words, every activation command will start or restart the snapshot function; a single deactivation command switches off the snapshot function completely.

By performing a regular status query via AMO-SDSU, users can determine which peripheral units are excluded from alarm monitoring, i.e. the new NOAL status in the status of the affected unit indicates whether or not the unit is excluded.

IMPORTANT: Further information will be available in the section entitled "AM Handling of Alarms" in the service manual.

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10 Country-Specific Features

Basically all the features in sections 2 to 10 are also available for those countries for which a release has been granted.

There are exceptions with respect to certain system configurations due to

1. the interface availability in the countries concerned (e.g. ISDNcentral-office interfaces),
2. the foreign-language user interface, and
3. approval regulations in the various countries.

10.1 Austria

10.1.1 Call Processing Features

10.1.1.1 Re-Ring from the Long-Distance Exchange

When the internal user has gone on-hook, the connection is not released in the exchange but forcibly released to the attendant through grounding of the b-wire on the exchange side.

10.1.1.2 Attendant Intercept of all Unsuccessful Connections

All unsuccessful connections due, for example, to:

- user busy,
- incomplete dialling,
- user free but does not answer,
- All Trunks Busy (ATB),
- user not suitably classmarked,

are forcibly released to the attendant position.

10.1.1.3 Redirect Function to Night Answering Position for Incoming DID Calls

External callers who reach a busy user of a PABX system switched to night service have the option of redirecting the call to the attendant or night answering position of the PABX.

10.1.1.4 Auto-Override by PABX System for Incoming DID Calls

For incoming DID calls to a busy user of a PABX system switched to night service, the system can automatically override the busy user's call. During the override procedure the internal user receives the override tone from the PABX system.

10.1.1.5 Flash Function to the Central Office

The flash function to the central office allows the user to employ certain public services available to him. The services are selected by means of suffix dialling. The only service currently offered by the central office is "tracing an external party".

10.1.1.6 Tracing an External Party

This service is activated with the flash function (i.e. without suffix dialling). This prevents that the connection to the external subscriber is released by the exchange.

10.1.1.7 Personal Calls to the Attendant Console

If the attendant's headset is not plugged, incoming inward-dialled calls from the exchange to the attendant's personal telephone number are immediately re-routed to the next attendant console or the night service station.

10.1.2 Analog Trunk Calls

Bothway Trunk Calls for Supervisory Frequency System (UEFS) with DID (TMAS8)

Two-wire main station line to the central office for voice and signalling with the following features:

- dc-transmissive lines

- basic signalling employing the loop procedure
- feeding in the central office (60V ±4V with 2x (359 < RSP < 690) Ω)
- emergency operation
- Out-dialling signalling system for outgoing exchange calls: DTMF
- Call charge pulse frequency for outgoing exchange connections: 12 kHz

The line can be operated on a

- bothway
- o/g only
- i/c only

basis.

- The following criteria are detected in incoming traffic:
  – line circuit seizure (12-kHz pulse)
  – direct inward dialling criteria (12-kHz pulses)
  – forward release (exchange feed voltage interrupt)
- The following criteria are transmitted down the line in incoming traffic:
  – answering criterion (loop closure in the PABX)
  – end-of-dialling information received
  – flash signal
  – backward release (by opening PABX loop)
- Emergency operation: seizure with 50-Hz ringing (as in dial system W48)
- Answering and call state essentially the same as in dial system W48
- The following criteria are transmitted down the line in outgoing traffic:
  – initial seizure (loop closure in the PABX)
  – DTMF dialling signals
  – flash signal
  – forward release (by opening PABX loop)
- The following criteria are received in outgoing traffic:
  – exchange dialling tone as seizure acknowledgement
  – call charge pulses from exchange (12-kHz pulses)
  – backward release (exchange feed voltage interrupt)

- Outgoing direction:
  Outgoing traffic is the same as for Switching System W 48. This therefore also permits use as an o/g trunk circuit for W 48 (except with simultaneous 50-Hz call charge data transfer), especially in the case of systems for i/c trunk calls with DID from the group selector.

10.1.3 Bothway Tie Trunk with Loop Reversal, Feed or Loop in Idle State for Pulse Dialling and DTMF Signalling (TMLR)

Two-wire, bothway DID tie trunk to the distant system for voice and signalling.

Signalling takes place using the loop procedure with

- feeding in the idle state with/without line supervision
- loop in the idle state on the line.

Signalling method: pulse dialling or DTMF

Features

1. With feeding in the idle state
   - Incoming traffic
     - Seizure with/without line supervision
     - Start-of-pulsing (optional)
     - Dialling
     - End-of-dialling (optional)
     - Answering (Option)
     - Forward release
     - Backward release
   - Outgoing traffic
     - Seizure with/without line supervision
     - Start-of-pulsing
     - Dialling
     - End-of-dialling
     - Answering
     - Forward release

– Backward release

2. With loop in the idle state

In contrast to the version with feeding in the idle state, the line supervision, start-of-pulsing, end-of-dialling and answering criteria do not apply.

10.1.4 Audible Tones for Austria

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone	425	Continuous tone or **300**/240/**1200**/660
Busy tone	425	**220...400**/200...400
Internal dial tone	425	**200**/300/**200**/300/**200**/800

The following audible tones are to be sent by the PABX::

Tone	Tsl	Frequency [Hz]	Pulse / Pause [ms]	Level [dBm0]
Internal dial tone	1	425	**200**/300/**200**/300/**200**/800	-3.4
Exchange dial tone	2	425	Continuous tone	-3.4
Ringback tone	4	425	**1000**/4000	-3.4
Busy tone	5	425	**300**/300	-3.4
Override tone	6	425	**200**/300/**200**/1300	-10.4
Call waiting tone	7	425	**100**/1900	-8.4
Data call tone	9	1300	**600**/1800	-0.4
Special dial tone	3	425+400 superimposed	Continuous	-3.4
NU tone	8	950/1400/ 1800	Triple tone **320-320-320**/960	-3.4
Test tone (Austria only)	15	700	Continuous	-4
Conference auxiliary tone 2	23	425	Continuous	-10.4
Conference auxiliary tone	24	425	Continuous	-8.4
LCR	26	1800	**340**/200/**340**/200/**340**/1000	-3.4

Special tone sequences:

Tone	Sequence	derived from
Conference tone	200 ms 300 ms 200 ms Continuous	Conference auxiliary tone 2 Pause Conference auxiliary tone 2 Silence
Call waiting tone	Continuous	Call waiting tone
Call waiting tone for night service terminal	Continuous	Call waiting tone
Override tone	Continuous	Override tone

10.2 Belgium

10.2.1 Analog Trunk Calls

Incoming Trunk Calls with DID for MFC-R2 Signalling with Loop-In Signal Loop (TMANI)

Two-wire main station line for voice and signalling

Signalling method:MFC-R2

The basic signalling is the loop procedure with feeding from the public exchange. Feeding is only supplied from the PABX in the blocking condition in the case of systems with potential blocking.

Signalling is by loop interruption in keeping with the operating mode "blocking".

Features:

- Seizure
- Dialling
- Answering
- Forward release
- Backward release
- Blocking

Bothway Main Station Line with Pulse Dialling and DTMF (TMANI)

Two-wire main station line to the central office for voice and signalling.

The line can be operated on a

- bothway

- o/g only
- i/c only basis.

Signalling method: pulse dialling or DTMF.

The i/c traffic always results in forced release to the attendant position.

Features:

- Incoming traffic
  – Seizure (ac ringing voltage)
  – Answering
  – Forward release
  – Backward release
- Outgoing traffic
  – Seizure
  – Dialling
  – Forward release
  – 16-kHz call data reception

The 1A procedure must not be used in Belgium.

The 1st trunk dial tone is basically injected at the user's side on a simulated basis. Although the 2nd trunk dial tone is evaluated by the PABX, it is not extended to the user.

10.2.2 Digital Trunk Calls

Only one-way operation is possible (i/c only with DID/o/g).

Incoming Call with DID and MFC-R2 Signalling (DIU-CAS)

Signalling method: MFC-R2

Features:

- Seizure
- Ready-to-dial state from the PABX
- Dialling
- Answering from the PABX
- Forward release

- Backward release
- Blocking from the PABX
- Register callback

Outgoing Call with DTMF (DIU-CAS)

Signalling method:DTMF

Features:

- Seizure
- Ready-to-dial state from the central office
- Dialling
- Answering from the central office
- Metering pulse from the central office
- Forward release
- Backward release
- Blocking from the central office
- Register callback

The 1A procedure must not be used in Belgium.

The 1st trunk dial tone is basically injected at the user's side on a simulated basis. Although the 2nd trunk dial tone is evaluated by the PABX, it is not extended to the user. The dialled digits are transmitted without delay. End-of-dialling is simulated to the system after a timeout (4 s in the case of DTMF users).

10.2.3 Analog Tie Traffic

10.2.3.1 Bothway Tie Trunk with E&M (TMEW2)

The signals are transmitted with a DC voltage via the E and M control wires.

50-Hz pulses or DTMF can be used as alternatives for dialling.

10.2.3.2 Bothway Tie Line with CEPT-L1 (TMSVF)

Four-wire access line for voice and signalling in the inband procedure (via TMSVF).

Signalling method:pulse dialling or DTMF

Features:

- Seizure
- Seizure acknowledgment
- Dialling
- Answering
- Releasing

Special criteria:

R2 signalling is not possible with tie traffic.

10.2.3.3 Analog Extension

Signalling is via a two-wire access line.

50-Hz pulses are used for main signalling.

DTMF is used for secondary signalling and dialling.

Signalling method: DTMF

10.2.4 Audible Tones for Belgium

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
1st exchange dial tone	425	Continuous
2nd exchange dial tone	900/1000/1140	**388**/388/388
ATB tone	425	**250**/250
Busy tone	425	**500**/500

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse / Pause [ms]	Level [dBm0]
Exchange dial tone	2	425	Continuous	1
Ringback tone	4	425	**1000**/3000	-1
Busy tone	5	425	**500**/500	-1
Override tone	6	425	**200**/3500	-8
Call waiting tone	7	425	**100**/1900	-8
Special dial tone	3	425+400	Continuous	-0.8
ATB tone	11	425	**240**/260	-1

Tone	Tsl	Frequency [Hz]	Pulse / Pause [ms]	Level [dBm0]
Positive acknowledgement tone	10	425	**40**/40	1
Internal dial tone	1	425	**2000**/300	1
NU tone	8	950/1400/ 1800	Triple tone **320-320-320**/1000	-8
2nd exchange dial tone (simulated)	13	900/1000/ 1140	**380**-**380**-**380**	1
Conference auxiliary tone	24	425	Continuous	-8
LCR	26	1800	**340**/200/**340**/200/**340**/1000	-1

Special tone sequences:

Tone	Sequence	derived from
Conference tone	200 ms 3500 ms 200 ms Continuous	Conference auxiliary tone Pause Conference auxiliary tone Silence
Call waiting tone	Continuous	Call waiting tone
Call waiting tone for night service terminal	Continuous	Call waiting tone
Override tone	Continuous	Override tone

10.3 France

10.3.1 Call Processing Features

10.3.1.1 Attendant Camp-On

When answering internal and external, incoming and outgoing connections on the attendant console, a 5 s timer is started. This time cannot be administrated and is activated via the country code.

During this time it is possible at any time to

- forward the connection

During this time it is not possible to

- hold or park the connection both by means of selective holding or parking and by automatic holding, e.g. in the case of erroneous operation;

- forward the connection after answering a recall.

While the timer is activated, pressing a key is ignored on the attendant console and a corresponding display appears.

10.3.1.2 Ringback Tone for Waiting External Party

If a waiting trunk enters the state "Calling an internal party", ringback tone keeps being applied to the waiting external party.

This covers the following situations:

- ringing state after switching,
- ringing state after a busy party became idle to whom
  - a call has been extended or
  - a call has been transferred in consultation hold.
- recall on the attendant console,
- ringing state after transfer from the party consultation hold.

During this waiting period the external party hears ringback tone.

10.3.1.3 DID to Busy User

The following can user-specific be set up for DID to a busy user (first-time busy):

- Attendant camp-on - busy station (check) for 15 to 40 s. The time can be set by means of administration and maintenance. When this time has expired, the call is released to the attendant console.

  The following applies to a Digite:

  - Only the first call waiting tone can be heard in the handset.
  - The subsequent call waiting tones result in signal on the piezo transducer and display.

  The following features can be defined per station for inward-dialled external calls if the station is busy (in talking state):
  During the waiting time,

  - external (exchange) callers receive the ringing tone.
  - internal callers with analog telephones receive the camp-on knocking signal.
  - internal callers with digital telephones receive an alerting tone on the piezo transducer and a display output.

- hunting group members all receive the appropriate signal, depending on their telephone type.

- Immediate attendant intercept

In the case of DID to busy user for whom a waiting call has already been entered busy tone is signalled to the exchange (no intercept, no second override).

10.3.1.4 Waiting Calls in a Hunting Group

The number of calls waiting in the call queuer of the hunting group equals the number of existing stations in the hunting group.

The number of stations changes when individual stations are removed or added.

A call for a hunting group is camped on at all stations of the hunting group. The first idle user of the group receives the call.

If several calls are waiting in the hunting group, the call waiting longest is assigned to the first idle user.

In a hunting group, the number of waiting calls can never be greater than the number of users.

10.3.1.5 Supervision and Visual Display of the Trunk State (Idle/Busy) with ELSD

This function is performed by the Exchange Line Status Display ELSD facility, which is physically separated from the trunk interface.

- Analog trunks
  An integrated solution will not be implemented.
- Digital trunks
  The state supervision takes place via administration and maintenance either by interrogation or by traffic measurement.

10.3.1.6 Second Dial Tone

Recognition of 2nd dial tone
The digit will not be outpulsed until the 2nd dial tone has been reliably recognised.

Connection
If the memory contains no digits, the internal user receives the simulated 2nd dial tone up to the next selection on the digit position where the trunk connection waits for the 2nd trunk dial tone.
The 2nd dial tone is not applied when speed calling or repertory buttons are used, if further digits are stored and with number redial.

10.3.1.7 Consultation Hold - Special Case

If the consulting user does not answer within 40 s after going on-hook in consultation hold and a resulting recall (immediate or timed recall)

- a user held internally will be released or
- a user held externally is intercepted to the attendant. The originator is called for up to 40 s before intercept takes place.
  - Attendant calls (but not calls to the attendant's personal telephone number) via lines with backward release are released after 3 minutes.

10.3.1.8 Waiting at a Station with Override Security

Waiting at a station with override security is not allowed.

When dialling a busy user with override security, the following happens:

- An external party is intercepted to the attendant.
- An internal party, the attendant as well as incoming tie traffic connections receive busy tone.

10.3.1.9 Recording Inadmissible Connections

The recording of inadmissible connections (ITR matrix) can be checked by the postal authority.

In order to use systems in places where they can be shared by several companies, selective barring of

- internal connections and
- transit connections

as well as selective access to

- optional remote service lines,
- special lines and
- dedicated lines

is possible.

10.3.1.10 Call Waiting Tone after Extending a Trunk Call to a Busy User

If an incoming trunk call is extended from the attendant to a busy user, this user receives the call waiting tone.

- Frequency 440 Hz
- Transmit rhythm
  - Tone 300 ms
  - Pause 20 - 40 s (can be administered).

In the case of camp-on to a hunting group, the tone is only applied to at least one member.

10.3.1.11 Special Dial Tone

If a user cannot answer the calls directed to him, e.g. because call forwarding or "Do-not-disturb" is activated, he receives special dial tone when uncradling the receiver.

10.3.1.12 Delayed Ring Tripping

Some trunk facilities provide delayed ring tripping in the case of incoming trunk calls. Provision has been made that the exchange does not hear the ringing current when answering.

10.3.1.13 Priority/Overload Display at the AC for Incoming Trunks

Overload statuses are visually indicated on the attendant console.

The overload LEDs on the attendant console keyboards are used exclusively for overload signalling.

The overload signalling feature is divided into three categories:

- All attendant consoles are busy; an exchange call is waiting for one of the attendants to go on-hook. In this case, the overload LEDs on all the attendant consoles in operation flash slowly.
  The LEDs do not flash rapidly unless one of the situations described under points 2 or 3 occurs.
- The number of waiting exchange calls is greater than the number of attendant consoles in operation.
  In this case, the overload LEDs on all the attendant consoles in operation

flash rapidly.
Calls which have already been allocated to an attendant console are not considered to be waiting calls even if they have not yet been answered.

- An exchange call has to wait for longer than 20 seconds before being answered.
  Unless this call has already been allocated to one of the attendant consoles, the overload LEDs on all the attendant consoles in operation flash rapidly. If the call has already been allocated to one of the attendant consoles, only the overload LED of that attendant console will flash rapidly.
  Flashing stops when the call is answered.
  - Re-rings are only carried out at the attendant console which originally transferred the call. Other attendants cannot answer the call if the original attendant console is still in operation. If the re-ring continues for longer than 20 seconds, only the overload LED of the attendant console which originally transferred the call will flash rapidly.
- The overload LED function is also intended for centralised attendant services configurations.
- The waiting time (as described in point 3) can be configured via AMO, as of HiPath 4000. The timer can be set between 20 and 100 seconds (default = 30 s) with the ATNDTRNS parameter of the CTIME AMO. Each time the waiting time is exceeded by a waiting call, the overload LEDs will flash rapidly (see point 3).

10.3.1.14 Three-Way Conference Override

The feature "three-way conference override" (intrusion) is implemented for the central attendant console (not for standard users).

10.3.1.15 Call Forwarding Destination Status Check

A variable call forwarding destination will only be accepted if the destination station (attendant console, Anate, Digite) is "ready" and the do-not-disturb function is not activated. Alternatively, the HiPath system can also be configured not to carry out these checks until call forwarding is actually executed.

10.3.1.16 PCM Frame Signal Parameters for DIU-CAS/DIUT2

The national and international bits of the PCM frame signal can be assigned different parameters per PCM highway. This allows the highways to be configured for specific applications, e.g. such as required in France.

10.3.1.17 Exchange Re-Ring

Incoming and outgoing exchange calls (first-time calls and re-rings) rung at the attendant console, the night service station or at standard user stations are terminated after 3 minutes unless answered. The 3 minutes are timed from the moment the ringing tone is sent to the calling party. Each time the call status changes (forwarding on no answer, transfer/call offer), the timer is started anew.

This function is implemented internally and also network-wide, in the case of networked systems linked via CorNet-NQ lines.

10.3.1.18 Incoming Exchange Call Recovery during Soft Restarts

For France, the recovery parameters will also be set to "one-party save" or "call ref. save" via AMO. This ensures that exchange connections are saved during recovery as follows:

1. Incoming digital and analog exchange calls, which have already been answered by the called party (exchange has received answering criterion) and are no longer in a standard two-party connection state, are saved according to the "one-party-save" principle, i.e. only the status data of the exchange line is saved.

   These incoming exchange calls are re-rung at the attendant console after the soft restart has been completed.

   Incoming digital and analog exchange calls, which are saved according to the "one-party-save" principle, include the following connection types:

   - Attendant console connections in any state after answering, e.g.
     - Exchange line connections in talking state (exception, since these connections are currently not saved with the "two-party-save" principle)
     - Exchange calls on consultation hold
     - Exchange calls on hold
   - Exchange calls on consultation hold
   - Exchange calls participating in a three-party conference
   - Exchange calls in intrusion (override) state
   - Exchange calls with call waiting for internal station user (knocking)
   - Parked exchange calls
   - Exchange calls on toggle hold

2. Outgoing exchange calls, which are not saved according to the "two-party-save" principle, are terminated by the soft restart protocol.
3. Incoming or outgoing exchange calls in other call processing states, such as
   - connection setup state
   - connection cleardown state,

   are terminated by the soft restart protocol.
4. The "one-party-save" or "call-ref.-save" recovery only saves exchange connections. Tie-trunk connections are not saved; tie-trunk connections set up via exchange lines in the remote system report the soft restart to the exchange circuit via the message interface.
5. For internal network configurations, digital networking trunks are required for this feature. This is because some circuit release data cannot be transmitted to exchange trunks in foreign nodes via analog lines.

For incoming exchange connections to standard user stations (or special equipment), the "one party save" is only carried out if a "two-party save" has already taken place. Since incoming exchange calls to the attendant console are not saved via a "two-party save", these calls are always saved via the "one-party save" option.

In the HiPath, the internal connection paths are released during a soft restart. However, the exchange caller will remain connected to the incoming exchange trunk and will receive the standard waiting tone or announcement set in the system for such cases.

Exchange calls saved via "one-party save" are entered as attendant re-rings in the call queue when the HiPath system has recovered (i.e. all exchange calls are re-rung at the attendant console regardless of the original internal call path before soft restart). These re-rings are treated as transfer calls to the attendant console via consultation hold, and are signalled on the attendant's line key with the appropriate priority (VFGR AMO).

If possible, any internal call paths of exchange calls to or via an attendant console which existed before the soft restart should be re-constructed. If this is not possible (e.g. call was not connected via attendant transfer), the re-ring is assigned to the attendant console group entered in the same ITRGR as the incoming exchange trunk.

Exchange call re-rings to the attendant console which were saved according to the "one-party save" principle can be processed as normal calls, i.e. the full range of call processing features is available.

The "one-party save" principle only applies to "voice" connections, which includes the voice mail service.

Exchange calls saved according to the "one-party save" principle receive priority treatment after a soft restart. In order to make this possible, attendant consoles are given set-up priority in the recovery system for the periphery.

10.3.1.19 Automatic Call Setup on Timeout (Timer-Dependent Hotline)

French PTT requirements dictate that automatic call setup on timeout (timer-dependent hotline setup or off-hook recall) must only take place from the initial seizure state. As soon as a user with this feature has dialled the first call number digit, the off-hook recall function must be deactivated. This can be configured via AMO. In countries other than France, the feature is customarily configured to begin call setup after the inter-dialling pause timer has run out, regardless of whether the line circuit is in initial seizure state or dialling was begun.

10.3.1.20 Transition to Intercept State

In accordance with French requirements, this function ensures that two-party connections are terminated on both sides. If a two-party connection is terminated or interrupted, and one party remains on-hook, he or she will hear the busy tone for 5 seconds before being assigned an attendant intercept status and transferred to the attendant console. This applies regardless of the device types engaged in the two-party connection.

10.3.1.21 Blocking Parallel Ringing to Attendant Console and User Station

This feature meets French PTT requirements for attendant intercept transfers of direct-dialled incoming exchange calls to PABX users who do not answer within the configured time. These calls are rung at the attendant console only.

Parallel ringing is also blocked for users in networked systems with central attendant consoles.

Busy stations are not re-checked for an idle state.

Parallel ringing is deactivated (blocked) by setting the feature bit string via AMO.

Attendant intercept transfers of incoming DDI exchange calls to PABX users who do not answer within the configured time are signalled with the same priority as initial calls waiting longer than 20/40 seconds. Recalls are signalled with the priority assigned to recalls.

Calls transferred to the attendant console after timeout with attendant intercept status are indicated as attendant intercept calls on the attendant console display. Re-ring calls are indicated as re-ring calls.

Once the attendant answers an attendant intercept or re-ring call, the dialled, the reason for the attendant intercept transfer or re-ring is displayed, i.e. "NO ANSWER" if the station line circuit was idle and the station user did not answer within the configured time.

Once the attendant answers an attendant intercept or re-ring call, the dialled, the reason for the attendant intercept transfer or re-ring is displayed, i.e. "BUSY (EXTERNAL/INTERNAL)" if the station line circuit was busy and the station user did not go on-hook within the configured time.

In both cases, i.e. no answer or busy, line 11 of the attendant console display will contain the message "NOT SWITCHABLE" if parallel ringing is blocked.

As for standard parallel ringing, when the attendant answers an intercept or re-ring call, the user originally dialled is shown on the display. However, if parallel ringing is deactivated, the attendant cannot simply switch the call back to the station line circuit or override the user's connection if the line circuit is busy. The attendant must re-dial the number of the station user or dial a different user's number and transfer the call as for the "attendant connect" feature.

If parallel ringing is not deactivated (= countries other than France), the attendant can switch the call back to the station line circuit in waiting or override state. If the caller wishes to be connected to a different user, the attendant must first release the initial internal connection path by pressing the CLEAR key.

10.3.2 Analog Trunk Calls

Incoming Trunk Calls (similar to BPO) with SOCOTEL Register Signalling (TMLSF)

Two-wire main station line for voice and signalling.

Signalling method:MFC-SOCOTEL

Features:

- Seizure
- Forward release
- Availability
- Seizure check
- Answering
- Clearing signal
- Barring

The following signals are exchanged

A2, A3
B1, B3
b1 - b0

Bothway Trunk Calls (similar to main station interface) with Incoming Attendant Intercept (TMANI)

Two-wire main station line for voice and signalling.

Signalling method:pulse dialling or DTMF

Features:

- Seizure
- Dialling (pulse dialling or DTMF)
- Forward release
- Exchange feeding polarity change
- 1st trunk dial tone
- 2nd trunk dial tone
- TMANI
- 12-kHz call charge pulses

The line can be operated

- bothway,
- i/c only,
- o/g only.
- Outgoing traffic is fully automatic. The signalling methods used are pulse dialling and DTMF. Incoming calls are always intercepted to the attendant console.

Signalling

- For outgoing calls
  The following signals are transmitted:
  - Seizure
  - Dialling (pulse dialling or DTMF)
  - Forward release

  The following must be recognised:
  - Polarity change of feeding from the exchange (wire crossing) for:
    talking partner answer (answering from the exchange side)
    release from the exchange side

- 1st trunk dial tone
- 2nd trunk dial tone
- Call charge pulses 12 kHz

  Feeding interruptions of up to 250 ms are ignored. Polarity changes ≥ 400 ms are considered admissible. The change of polarity toggles, i.e. it has no defined initial state.

- For incoming calls:
  The following is recognised:
  - Seizure (AC ringing voltage)
  - Forward release

  The following is transmitted:
  - Answering
  - Backward release

End-of-dial control for pulse dialling and DTMF in accordance with the 1A signalling method.
Speech path through-connection is with the simulated end-of-dialling for outgoing calls.

Call charge pulses
Call charge pulses with 12 kHz are received.

First and second dial tone
The following dial tones are received, evaluated and simulated to the user in the case of outgoing trunk calls:
1st dial tone (IT1, for local calls). This dial tone consists of a 440-Hz continuous tone.
2nd dial tone (IT2, for long-distance calls). This dial tone consists of a continuous tone which is formed by superposition of two frequencies, i.e. 440 Hz and 330 Hz. Only 330 Hz is detected.

No specific treatment has been provided for the use of special services and additional features in the exchange.

10.3.3 Digital Trunk Calls

10.3.3.1 Incoming Trunk Calls (DID, similar to BPO) with SOCOTEL Register Signalling

Two-wire main station line for voice and signalling.

Signalling method:MFC-SOCOTEL

Features:

- Seizure
- Dialling: MFC-SOCOTEL
  - forward:
    b1 - 10 dial information
  - backward:
    A2 end-of-dialling
    A3 changeover to B-signal
    B1 user idle
    B3 user busy
- Forward release
- Idle state
- Answering
- Forward release by the exchange
- Backward release after answering by the PABX
- Barring by the PABX
  - No forward disconnection supervision in the PABX
  - PABX only bars in the idle state
  - Reseizure by the exchange not transmitted until 500 ms after the forward release. This time is not supervised in the PABX
  - No "Backward release before answering" line signal

Register signalling in accordance with SOCOTEL signalling. The following signals are exchanged:

A2, A3
B1, B3
b1 - b0

10.3.3.2 Bothway Trunk Calls (similar to Main Station Interface) with Incoming Attendant Intercept

The line can be operated

- bothway,
- o/g only,

- o/c only.

Outgoing traffic is fully automatic. The signalling methods used are pulse dialling and DTMF.

Incoming calls are always intercepted to the attendant console.

Two-wire main station line for voice and signalling.

Signalling method:pulse dialling or DTMF

Signalling

- For outgoing calls
  The following signals are transmitted:
  – Seizure
  – Dialling (pulse dialling or DTMF)
  – Forward release

  The following is recognised:
  – Idle state
  – 1st trunk dial tone
  – 2nd trunk dial tone
  – Call charge pulses
  – Answering
  – Backward release
  – Barring (outgoing and bothway)
- For incoming calls
  The following signals are transmitted:
  – Idle state
  – Answering
  – Backward release
  – Barring (bothway and outgoing)

  The following is recognised:
  – Seizure (causes attendant intercept)
  – Forward release

Signalling takes place isochronously with the analog HKZ signalling via bits a and b of the 16th time slot in the pulse frames 1 - 15.

In the case of forward release of outgoing calls after answering, all call charge pulses are completely evaluated up to the release performed by the exchange.

After release, the line is barred for outgoing seizures for 500 ms, irrespective of the previous direction of seizure. No provision has been made for longer periods.

1A signalling is used for end-of-dial control for pulse dialling and DTMF.

Speech path through-connection for outgoing calls takes place with the simulated end-of-dialling.

If an incoming seizure arrives during the glare time of an outgoing seizure,

- the outgoing seizure is cancelled,
- the system is switched to the idle state,
- the incoming seizure is intercepted to the attendant console.

Answering is immediately recognised for outgoing calls. Thus, the output of further digits is prevented. Dialling further digits can be interpreted by the exchange as release.

While transmitting forward release, an incoming seizure is ignored until the incoming exchange side sends idle signal (availability) again.

The call charge pulses are signalled via bit b.

No specific treatment is provided for the use of special services and additional features in the exchange.

Line signalling (outgoing):

- Only 50/50 ms dial pulses are permissible (66/33 ms).
- Call charge pulses which are too long or too short are ignored.
- A call charge pulse is ignored if the length of the pause since the previous pulse or the answering criterion is <75 ms.
- Call charge pulses remain valid up to the backward release.
- If a call charge pulse is present at the time of the backward release, it is evaluated.
- The backward release is not timed.
- The 1st and 2nd trunk dial tone is detected.
- A backward release is acknowledged immediately by the PABX with a forward release.
- The protection time for outgoing seizures starts when the PABX transmits the idle signal.

- When the exchange answers, the dialling output is aborted immediately, since otherwise the exchange might interpret continued dialling as a release procedure.
- The speech path is not switched for outgoing traffic until the end-of-dialling is simulated after a fixed period in the DH or after answering.
- The timer for simulating the end-of-dialling can be set on a device-specific basis.
- The second dial tone must be detected after digits 16 and 19.
- If an incoming seizure is received during the glare time of an outgoing seizure
  - the outgoing seizure is returned to the exchange and
  - Intercepted by the attendant.

    It is not possible to establish whether the exchange has interpreted the outgoing seizure as answering followed by release, or whether it has ignored it. The seizure is not intercepted by the attendant until after expiry of a fixed period, in order to prevent phantom calls at the attendant console.
- Barring by the PABX is always signalled to the exchange, and is initiated in the idle state. No distinction is made between one-way and bothway lines.
- Barring by the exchange can be detected after a fixed period in the idle state and after a forward release.
- The 1A method is used for DTMF users and DTMF trunks.

Line signalling (incoming):

- A backward release is possible after answering. The release by the exchange is not timed.

10.3.3.3 Bothway Trunk Calls with ISDN Exchange Circuit T_2 (Numeris Line)

Four-wire main station line for voice and signalling. Implemented in accordance with ISDN (RNIS) protocol VN4. Interface to DIUT2.

The following features are used by HiPath:

- Basic call setup and cleardown
- Direct inward dialling (DID)
- Identity of the calling end user
- Identification of the calling subscriber
- Subaddressing

- Calling end user identity secrecy
- Current call charge information (telecharging)
- Call charge information at the end of call (total cost)
- Emergency connection (essential line connection)
- Access with dedication
- Temporary terminal transfer
  Implemented via 2nd B channel, not an ISDN public network exchange feature.
- Nuisance call identification

The line can be operated

- bothway,
- o/g only,
- i/c only.

10.3.4 Overview of all Numeris Supplementary Services

- Routinely offered services

  These services are (routinely) offered to all users. They can therefore also be regarded as a constituent part of the basic procedures. No registration or cancelation procedures are required for these services.

  – Portability during communication (not relevant at T reference point)
  – Identification of the calling subscriber
  – Subaddressing
  – Call waiting (not relevant at T reference point)
- Other supplementary services
  – User-user service 1
  – Essential line connection
  – Access with dedication
  – Line connection with restriction
  – Calling end user identity secrecy (no)
  – Identity of the calling end user

- Telecharging
- Total cost
- Temporary STI transfer (not relevant at T reference point)
- Temporary terminal transfer
- On hold and dual outgoing calls → only provided for the basic access (not relevant at T reference point)
- Transfer → only offered by NT2 (nicht relevant am T-Referenzpunkt)
- Add-on conference → only offered by NT2 (not relevant at T reference point)
- Nuisance call identification
- Direct inward dialling → only provided as a supplementary service at Stage 1
- Transgroupe (Basic Call Setup)

Numeris Supplementary Services used in HiPath 4000

Identification of the calling subscriber

In the setup message the network supplies the "Installation Designation Number (IDN)" of the calling subscriber with the I element "Origination Address".

A precise definition of the IDN is contained in the the T22 specification for "ISDN Adaptation France".

Subaddressing

The network supports the transparent transmission of a max. 4-digit subaddress. Interpreting the meaning of the subaddress is a matter for the terminating terminal.

Note: The method described in the Numeris specifications, with the subaddress serving within the PABX to address the extensions, is not employed in HiPath.

Direct inward dialling (DID)

In Numeris specifications RSM/133 and STP/1913, direct inward dialling is not described under the supplementary services but under Basic Call. This thus makes it possible for the last 4 digits of each destination address to be transmitted to the PABX as a block. This limitation with Numeris does not allow DID numbers or flexible numbering schemes extending beyond these 4 digits.

Essential line connection

With this service, outgoing calls are given a higher priority than incoming calls. It is only allocated administratively. The service has no particular effect on D channel signalling, i.e. it is automatically taken care of with the Basic Call.

Access with dedication

This service permits specialisation of a certain number of B-channels of a user interface (raccordement d'usager).

The B-channels can be subdivided into

N:	Number of B channels per user interface
N1:	Number of outgoing B channels
N2:	Number of incoming B channels

Calling end user identity secrecy

This service prevents called parties from being given the calling party's identity. It can only be allocated administratively, after which it is active for all calls from the relevant terminal. Individual activation of this service for each call is not possible with Numeris.

Important: When the "Calling end user identity secrecy" service has been activated, only the A-number section supplied by the central office (IDN) is masked out. There is no suppression of the A-number section (SDN) that may be additionally supplied by the subscriber.

This feature is used for outgoing connections from the HiPath only in call-by-call mode.

Identity of the calling end user

Independently of the identity of the calling end user (supplied by the exchange), the A-party can additionally provide his own identification in the setup message in an "Origination Address" I element. The network transmits this I element end-to-end without taking note of it.

Telecharging

This service supplies the subscriber with the current charge data (real-time charge units). There are two versions of this service in Numeris:

All outgoing calls of a line have this service administratively allocated (contracted).

Total cost

In contrast to the "Telecharging" service, this service only provides the subscriber with the total call costs at the end of the call. The service can only be contracted and so does not require any activation/deactivation procedures in the D channel.

Temporary terminal transfer

This service differs from the "Temporary STI transfer" service in that each extension can be transferred individually and not just the entire Numeris line connection or the entire PABX.

The service corresponds in character to the ITU service "Call deflection". A transferred terminal B acknowledges a setup message with the RELEASE message, in which it also indicates its transfer destination C. The exchange uses this transfer address to set up the A-C connection again.

The Numeris service "Temporary terminal transfer" is in competition with the HiPath service "Network-wide call forwarding".

Advantage of the Numeris service:

- No B channel is occupied to the PABX (compared with 2 B channels in the HiPath service).

Disadvantage of the Numeris service:

- The call charges for the sections A–B and B–C of the connection are calculated and registered separately in the exchange. However, transferring user B in the PABX is not informed of the call charges for section B–C (→ consistency problems between call data registration in HiPath and in the exchange).
- A charge is made for the Numeris service for setup and for each application.

Nuisance call identification

This service allows the user to have the data (A-No., date, time,...) of a nuisance caller registered by the network and printed out while a call is in progress or during the release phase. Nuisance call identification has priority over the service "Calling end user identity secrecy".

Line connection with restriction

This service is offered by the HiPath service with much greater convenience with the EWAKO. The main advantage compared with the Numeris service is that the connection possibilities (toll accesses) can be set up individually **per user** with a COS (class-of-service). With Numeris this is only possible for the entire Numeris line connection or the entire PABX.

User-user service 1 (UUS1)

This service allows terminals and the PABX to exchange private information in the D channel during the connection setup and cleardown phase. For this, Numeris has defined the information elements

UUI	User - to - user information (for TE) (user-side)
UUF	User - to - user facility (for PABX) (exchange side)

Transgroupe

The Transgroupe service allows geographically separate private networks to be grouped together as one corporate networked system with a corporate call charge tariff system.

Transgroupe connections to remote systems are set up without the standard exchange access code (usually "0"). User access authorisations (dialled number restrictions) are evaluated on the basis of a new class-of-service yet to be defined. Dialled numbers which contain a leading "0" are correspondingly evaluated as calls into the public exchange network.

Access to a transgroupe user from a standard network (non-Transgroupe user) is controlled via the "barred/unbarred access" feature in the public network. "Barred access" callers receive a DISCONNECT message down the line. "Unbarred access" calls are set up as normal.

The following dialled number evaluation procedures (digit analysis) are defined in the Numeris specification for the Transgroupe feature:

- In partial digit analysis (standard procedure), only the first few digits of the Transgroupe number are evaluated, in order to identify the destination system. The remaining digits are identical to those entered in the public network dial plan.
- Full digit analysis allows complete independence from numbers entered in the public network dial plan.
- Multiple, full digit analysis enables a connection setup between a user station number in the private network dial plan and several different subscriber numbers in the public network, depending on the call originator.

The HiPath system only supports the digit analysis procedures decribed in points 1.) and 2.).

Numeris Supplementary Services not used in HiPath 4000 (negative list)

Temporary STI transfer

Due to the limited application range, HiPath does not employ this service but the much more interesting service "Call forwarding at the terminal" instead. This service is also allowed for single basic access interfaces.

On hold and dual outgoing calls

On hold and dual outgoing calls (consultation) are services that are implemented internally in HiPath 4000 (only for PBX users).

Transfer

In HiPath, this service is implemented locally with the features "Transfer during call" and "Transfer during ringing" (only for PBX users).

Add-on conference

Conferencing is offered by HiPath as an internal service. For this reason it serves no practical purpose to use the Numeris service in parallel with it (only for Communication Server users).

Portability during communication

This service is only relevant for an S_0 reference point.

Call waiting

HiPath does not make use of the call waiting service, i.e. a connection request with a SETUP without B channel is initiated by HiPath 4000 via protocol measures.This situation seldom occurs for T2 connections. If B-channels are free, these are indicated at the terminal as enabled for second call acceptance.

One-way B channels

This service enables a certain number of B channels of an S2 line to be configured for one-way traffic. If a number of B channels are configured for outgoing traffic only, the rest can be configured for incoming traffic only. This service can be assigned separately per B channel. The following parameters are administered in the system:

N	=number of B channels per connection
N1	=number of B channels for outgoing traffic
N2	=number of B channels for incoming traffic

Toggling

The Numeris service only applies to basic access interfaces at the S reference point. Toggling between an active and a held call is an internal HiPath service.

Packet mode

User data is transmitted via the D-channel of the X.25 interface.

Call forwarding activation during ringing

This Numeris service is not supported in the HiPath 4000 system. External callers are forwarded by means of the forward-on-no-answer service in HiPath.

User-to-User signalling service 2 (US2)

This service allows the transfer of user information messages between users during connection setup and disconnection.

10.3.5 Digital Tie Traffic

Bothway Tie Trunk with E&M Signalling

E&M signalling (RON/TRON) has been implemented through bit a of the 16th time slot in pulse frames 1 - 15 (DIU-CAS).

Bothway Interface to the Colisee Numeris Public Network (DIUT2)

For networking PABXs in France, direct access to the public Colisee Numeris network is possible with the following criteria:

- Digital, bothway 2-Mbit/s interface via T2 (DIUT2)
- Signalling via D-channel as per VN3

Bothway Connecting Lines with CorNet-NQ Signalling via DIUT2

Linking networked HiPath systems with the CorNet-NQ protocol.

Criteria:

- Digital, bothway 2-Mbit/s interface via T2 (DIUT2)
- Signalling via D channel as per VN3

10.3.6 Signalling Method

The pulse dialling and DTMF dialling methods as well as the register signalling method MFC-SOCOTEL are used.

Pulse dialling is used

- in tie lines analog and digital, incoming and outgoing
- trunks (similar to MSI) analog and digital, outgoing.

DTMF is used with trunks (similar to MSI) analog and digital, outgoing.

MFC-SOCOTEL is used with trunks (similar to BPO) analog and digital, incoming.

The following signals are exchanged:

A2, A3
B1, B3
b1 - b0

10.3.7 Announcements and Tones

Audible tones

For reasons of uniformity, the PABX must always transmit the audible tones normally sent from the public exchange after the trunk code has been dialled. The audible tones simulated by the PABX are absolutely identical to those of the public exchange.

Announcements

- Announcements in French

- Music on hold
  The music/voice announcement alternatively available is generated in the system or in an external device (Musiphone). The corresponding monitoring functions of the connected external device are provided by the system operator.

10.3.8 PCM-30 Characteristics

1. No signalling in the following error cases:

- no signal,
- frame loss,
- alarm signal (SIA 2M9 from the far end = AIS),
- loss of multiple frame alignment word (PVMT),
- alarm display (SIA 64 k) from the far end.
- After a maximum of 30 ms, received signalling is not evaluated.

2. Excessive error rate in the pulse frame alignment word

- Criteria for releasing the error display
  – For an error rate $\leq 10^{-4}$, the probability of releasing the error display within <5 s must be $<10^{-6}$.
  – For an error rate $\geq 10^{-3}$, the probability of releasing the error display within <5 s must be >0.95.
- Criteria for deactivating the error display
  – For an error rate $\geq 10^{-3}$, the probability of deactivating the error display within <5 s must be $<10^{-6}$.
  – For an error rate $\leq 10^{-4}$, the probability of deactivating the error display within <5 s must be >0.95.

    CRC4 is not supported.

3. Connection compatibility

It has been ensured that switching equipment can be connected to public exchanges even in the transition period.

In the following, the various compatibility cases are shown as well as the role of the respective connection ends.

The following four configurations will be dealt with:

- Digital connection to a public time multiplexing exchange 2G or to a TNE 2G.

- Digital connection to a public time multiplexing exchange 1G.
- Digital connection to a digital remote equipment 1G (version TNE 1a).
- Digital connection to a digital remote equipment 1G (version TNE 1b).

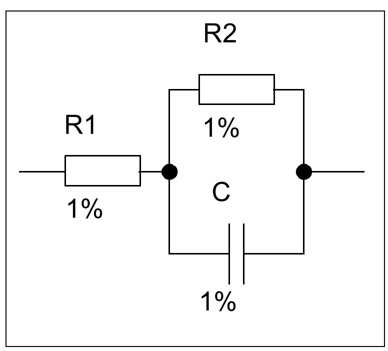
Parameterisation is possible by means of administration and maintenance.

10.3.9 Transmission Values

For the analog interfaces, the requirements according to ST/PAA/TPA/STP/1024, issue 4 dated October 1985, have been met. The transmission quality of the trunks can be checked from the exchange using the 800-Hz test.

1. Impedances (applies to analog interfaces only)
   - Zi interface

     It must be adapted to the telephones to be connected (which meet the postal standards) in the best possible way. If this is the case, the impedance is also suitable for the connection of voice frequency modems.
     The following network with the specified values is an arrangement with the impedance as requested for 1000 Hz:



Variant	R1[Ω]	R2[Ω]	C [nF]	Z [Ω]
1	180	910	150	836
2	213	1000	137	932
Variant 1 is standard, Variant 2 is the target				

   - Ze interface

     The impedance can have the following nominal values:

     – variant 1 of Zi

     – variant 2 of Zi

     – 900 ohms

   – 600 ohms

- Z1S4 interface

  No special requirements.

2. Level (applies to analog interfaces only)

   No special requirements must be met for digital interfaces

Interface	Level [dBr]	
	Input (A→D)	Output (D→A)
Zi	0	-7
Ze S (standard)	-6	-1
Ze C (short line <2 dB)	-3	-4
Z1S4	-3,5	-3,5

10.3.9.1 Subscriber Line Module, Analog

Adaptation facilities for constant current feeding and loop interruption

In the phases in which flash is expected loop interruptions between 190 and 350 ms are evaluated as flash.

A loop interruption of max. 350 ms is not evaluated as release. Any loop interruption >500 ms is evaluated as release.

Ringing cadence

- 1.5 s ± 5% ring
- 3.5 s ± 5% pause
  The ringing cadence can be parameterised via AMO.

Automatic dialling facilities which use crossed wires for operation in the public network cannot be connected to the SLMA.

10.3.9.2 System Characteristics

Power supply

In the case of battery operation, the battery allows interruption-free operation of at least 8 hours. Batteries do not form part of HiPath 4000.

Electrical safety according to the following standards:

- EN 60950
- EN 41003.

Electromagnetic compatibility according to the following standards:

- NF C46 - 021
- NF C92 - 130
- NF C98 - 020
- STPA - 1311

Duplex operation

For configurations with more than 64 trunks, duplex operation is possible for the main parts of the system.

During duplex switchover, existing connections and the associated signalling will not be interrupted. This ensures stable subscriber connections (internal and external).

10.3.10 Audible Tones for France

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone 1	440	Continuous tone
Exchange dial tone 2	330+440	Continuous tone
Busy tone	440	**500**/500

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse / Pause [ms]	Level (dBm0)
Internal dial tone	1	330	Continuous tone	-8±2
Exchange dial tone 1	2	440	Continuous tone	-11±2
Exchange dial tone 2	13	330+440	Continuous tone	330Hz: -8±2 *) 440Hz: -11±2
Ringback tone	4	440	**1500**/3500	-11±2
Busy tone	5	440	**500**/500	-11±2
Override tone	6	440	**200**/200/**200**/1400	-11±2
Call waiting tone	7	440	**300**/30000	-11±2
Special dial tone	3	440/440+330	440 Hz:Continuous tone 330 Hz: **750**/750	330Hz: -8±2 *) 440Hz: -11±2
NU tone	8	440/330/440	**350**/350/**350**/500	-11
*) Level incrementation is 3 ±1dB				

Tone	Tsl	Frequency [Hz]	**Pulse** / Pause [ms]	Level (dBm0)
Positive acknowledgement tone	10	330	**50**/50	-8
Check tone	15	800	Continuous tone	-2
Conference auxiliary tone	24	440	Continuous tone	-11
LCR	26	1800	**340**/200/**340**/200/**340**/1000	-11±2
*) Level incrementation is 3 ±1dB				

Special tone sequences:

Tone	Sequence	derived from
Conference tone	200 ms 300 ms 200 ms Continuous	Conference auxiliary tone Pause Conference auxiliary tone Silence
Call waiting tone	300 ms 1000 ms Continuous	External dial tone Pause Call waiting tone
Call waiting tone for night service terminal	300 ms 1000 ms Continuous	External dial tone Pause Call waiting tone
Override tone	Continuous	Override tone

10.4 United Kingdom

10.4.1 Call Processing Features

Digit prefixing for direct inward dialling (DID)

Leading DID digits are not transferred from the exchange to the PABX and must be prefixed for each incoming trunk group of the digit sequence arrived.

Password for administration and maintenance 6 characters. Max. 3 wrong entries are permitted.

Barring DID lines

When individual trunk circuits (incoming trunk circuit with feeding to the exchange) are barred, feeding will be removed. This applies to all barring facilities possible:

- AMO
- DEP

- switch on module
- system failure (also power failure)

Emergency telephones on analog DID lines in the case of system failure
In the event of a HiPath system failure, failure transfer of specific analog DID lines is performed to special terminals. These terminals have a separate power supply (feeding to the exchange) and are commercially available.

Emergency telephone with ground key on MSI line in the event of system failure (MSI emergency operation)
Changeover of MSI line to analog telephone (with pulse dialling and ground key) for signalling ground start via b-wire (during emergency operation, trunk seizure via ground key).

Do-not-disturb with DID without attendant intercept
In the case of systems with DID without attendant intercept, attendant intercept must still be performed for users with "Do-not-disturb" activated.

Attendant override protection for lines seized by direct dialling.

Continuous tone for user not connected or not authorised
In DID, no ringback or busy tone is returned if party is not connected **or** not authorised. Continuous tone will be applied.

10.4.2 Analog Trunk Calls

Incoming Trunk Calls with Direct Inward Dialling (TMLSF/TMLRB)

Two-wire main station line for voice and signalling. Application for short and long lines.

- Pulse recognition for 7 to 12 pulses/s.
- Break must be recognised with 40 - 84% (with line lengths between 0 and 6 km!)

Signalling method: pulse dialling

Features:

- Idle state
- Seizure
- Dialling
- Answering
- Forward release
- Backward release

Bothway Trunk Calls without DID (TMANI)

Two-wire main station line for voice and signalling with loop calling guarded clearing (transmission bridge in the exchange). Used for both short and long lines.

- Feeding in the exchange
- Loop signalling
- Release by means of loop interruption of 600 - 900 ms
- 50-Hz call charge pulse

Signalling method: pulse dialling or DTMF

Features:

- Idle state
- Outgoing seizure
- Incoming seizure
- Ready-to-dial condition
- Dialling
- Answering
- End of call
- Forward release
- Backward release
- No direct inward dialling
- Signalling similar to main station interface (outgoing dialling)

10.4.3 Analog Tie Traffic

Bothway Tie Traffic with DC 5-A Signalling

Four-wire line for voice and two-wire line for signalling with E&M method. Used for short lines (TMEMW, TMEW2 from HiPath 4000).

- DC5-A: Fast connection setup
  – Tie traffic basically bothway
  – Barring with BUSY OUT (backward busy) to exchange or tie line

- The following signalling variants have been implemented:
  Immediate dial (without seizure acknowledgment)
  Delay dial (with seizure acknowledgment, start-of-pulsing signal)
- Sending and receiving the dialled digits in the case of lines with and without seizure acknowledgment/start-of-pulsing signal
- DTMF dialling is not coupled with the "End-of-dialling" feature
- No suffix dialling possible after "End-of-dialling"

Signalling method: pulse dialling or DTMF

Features:

- Idle state
- Seizure
- Delay dialling (with seizure acknowledgment, start-of-pulsing signal) or
- Immediate dialling (without seizure acknowledgment)
- Dialling
- Answering
- Forward release
- Backward release
- Transmission and reception of dialling for lines with and without seizure acknowledgment/start-of-pulsing signal
- DTMF dialling not coupled to "end-of-dialling" line signal
- Suffix dialling not possible after end-of-dialling

Bothway Tie Traffic with DC 10 Signalling (TMEW2)

Two wire line for voice and signalling according to the loop method. Used for long lines.

- DC 10: Conversion from DC 5 to DC 10 by means of adapter

Signalling method: pulse dialling or DTMF

Features:

- Idle state
- Seizure
- Forward/backward holding
- Delay dialling (with seizure acknowledgment, start-of-pulsing signal) or
- Immediate dialling (without seizure acknowledgment)

- Dialling
- Answering
- Forward release
- Backward release

10.4.4 Digital Trunk Calls

Bothway Trunk Calls with DASS2 Signalling (DIUN2)

Implementation with gateway

- Transmission:PCM30 2,048 Mbit/s
  (30 B-channels)
- Signalling interface:2-wire (transmit/receive)
- Signalling protocol:CorNet-NQ/DASS2 in the signalling channel (TSL16)

Features:

- Voice service
- Display of called/calling subscriber (calling/called line identification)
- Call data registration

10.4.5 Digital Tie Traffic

Bothway Tie Traffic with DC 5-A Signalling (DIU-CAS)

Four-wire line for voice and signalling according to the PCM30 method (2.048 Mbits). Used for short lines, transition to Mercury network.

- Signalling DC 5-A via DIU-CAS

Signalling method: pulse dialling or DTMF

Features:

- Idle state
- Seizure
- Delay dialling (with seizure acknowledgment, start-of-pulsing signal) or
- Immediate dialling (without seizure acknowledgment)
- Dialling
- Answering

- Forward release
- Backward release
- Transmission and reception of dialling for lines with and without seizure acknowledgment/start-of-pulsing signal
- DTMF dialling not coupled to "end-of-dialling" line signal
- Suffix dialling not possible after end-of-dialling

Bothway Tie Traffic with DPNSS1 Signalling

Implementation with gateway for basic call and called/calling line identification (see also

Features in accordance with BTNR 188:

- Basic call setup and cleardown (basic call)
- Display of calling/called subscriber (called/calling line identification)
- Callback when subscriber is idle again after busy condition (callback when free, HiPath: callback when busy)
- Override (executive intrusion, HiPath: override)
- Camp on (call offer, HiPath: camp on)
- Call diversion / transfer of ringing (diversion)
- Hold / consultation hold (hold, HiPath: consultation)
- Intercept and recall caused by line seized in incoming direction (redirection)
- Add-on conference (three party, HiPath: call transfer, three party conference)
- Transit traffic in the sense of transparent transfer of all codes of the defined UELMs (transit working, HiPath: user to user, HiPath system between DPNSS 1 systems)
- Alternating (shuttle)
- Call extension in the network (redirection executive intrusion)
- Ringback in mailbox

Features not implemented:

- AC cannot camp on
- AC cannot be member of an add-on conference
- In the case of personal calls from the AC, callback cannot be initiated
- Call diversion without transport and only via the path to the forwarded-to subscriber (no bypass)

Applications for DPNSS1 in HiPath 4000:

- Tie traffic, also multiple configurations of Communication Servers and in combination with S_0/S_2 interfaces (CorNet-NQ)
- Main PABXs/satellite PABXs
  - HiPath as satellite PABX (without AC)
  - HiPath as main PABX (with AC)
- Main PABX/main PABX configurations with a central AC are not provided

10.4.6 Signalling Methods

Pulse dialling (DP) and DTMF (DTMF4) are the signalling methods used for the various analog and digital signalling types.

10.4.7 System Characteristics

Electrical safety

Electromagnetic compatibility (EMC)

Blind attendant console

- AC with adaptation for Braille writing in English

10.4.8 Audible Tones for the United Kingdom

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone	350+440	Continuous tone
Busy tone	400	**380**/380
ATB tone	400	**400**/360/**220**/520

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse/Pause [ms]	Level (dBm0)
Internal dial tone	1	400+450	Continuous tone	-6
External dial tone	2	350+440	Continuous tone	-6
Special dial tone	3	440+480	**200**/300/**200**/300/**200**/800	-6
Ringback tone	4	400+450	**400**/200/**400**/2000	-11

Tone	Tsl	Frequency [Hz]	**Pulse**/Pause [ms]	Level (dBm0)
Busy tone	5	400	**380**/380	-14,5
Override tone	6	1400	**200**/1500	-10
Call waiting tone				
NU tone	8	400	Continuous tone	-14,5
Data call tone				
Positive acknowledgement tone	10	520/400	**340**/**340**/800	-6
ATB tone	11	425	**400**/360/**220**/520	-6
Negative acknowledgement tone	14	400/520	**340**/**340**/800	-6
Conference auxiliary tone	24	1400	Continuous tone	-10
LCR	26	1800	**340**/200/**340**/200/**340**/1000	-11

Special tone sequences:

Tone	Sequence	derived from
Conference tone	2000 ms Continuous	Override tone Silence
Call waiting tone	100 ms 10000 ms 100 ms Continuous	Override tone Pause Override tone Silence
Call waiting tone for night service terminal	100 ms 100 ms 100 ms 10000 ms 100 ms 100 ms 100 ms Continuous	Override tone Pause Override tone Pause Override tone Pause Override tone Silence
Override tone	Continuous	Override tone

10.4.9 DPNSS1/DASS2 Features

If a CorNet line is switched via a CorNet-DPNSS gateway (CDG), HiPath provides DPNSS1/DASS2 signalling to the exchange (for UK and other DPNSS markets). The CDG converts CorNet-NQ protocol signals to DPNSS1/DASS2 and vice versa.

10.4.9.1 DPNSS1/DASS2 Enlargements on Basic Access

The 2-Mbit/s lines can be configured to provide the required number of channels by the HiPath system and to allow the remaining channels to be used for external devices, e.g. digital drop/insert multiplexers. This allows the 2-Mbit/s lines to be used for voice transmission and for inward data transmission from external sources.

During recovery of DPNSS1/DASS2 lines, channel reset requests are transmitted by the CDG. Any channel whose channel reset request is not acknowledged is taken out of service for voice transmissions in the CDG. This channel is then used for external data transmissions only.

The local channel interface is kept open in case a reset request message is received from the remote end.

10.4.9.2 Call Back When Free (CBWF)

This feature is an improvement on the (network-wide) feature "call back on busy", in that it works over multiple DPNSS1 connections. This means that the feature is fully functional is any type of DPNSS network, regardless of whether it is a star configuration or a delta configuration. The feature now allows the configuration of several parallel lines via CDGs.

10.4.9.3 Call Back When Next Used (CBWNU)

This feature allows "call back on no answer" (network-wide) for users connected to HiPath networks via DPNSS lines. The feature enables a caller, whose call is not answered, to set a call back request to be set up immediately following the DISCONNECT message the next time the called party uses his or her telephone. This feature is fully functional is any type of DPNSS network, regardless of whether it is a star configuration or a delta configuration.

If the called user's station is equipped with a mailbox LED, the message notification takes priority over the call back request.

10.4.9.4 Centralised Attendant Services (CAS)

This feature enables a central attendant console or console group to be configured in mixed configuration CorNet/DPNSS networks. The centralised attendant services consist of the features described in the following three chapters, with the addition of the loop prevention feature and the route optimisation feature.

Synonym: central operator/attendant

The centralised attendant services group can be configured in the HiPath system, or in the iSDX or iSLX system of the network. Calls to the central attendant consoles can be switched via any desired combination of these Communication Server types.

The following basic features are available:

- Incoming exchange calls to remote Communication Servers are re-routed to the central attendant console.
- Re-routed calls can be transferred back to the remote Communication Server.
- Route optimisation is carried out for all calls which are transferred back to the original Communication Server by the central attendant console, i.e. network loops are avoided.
- Calls transferred by the central attendant console and not answered within the configured time are re-rung at the attendant console (does not work if the original Communication Server is an iSLX system and the CAS group is configured in an iSDX).
- A CAS attendant can override (intrude on) a call conducted by a remote Communication Server user.
- A CAS attendant can camp on (knock on) a remote Communication Server user's line circuit.
- A remote Communication Server station can be configured as the night service station for CAS attendants.
- Remote Communication Server attendants can be assigned the same attendant access code
- Centralised attendant services differentiates between exchange call and internal call signalling at the central attendant console
- Exchange calls transferred by the central attendant are signalled with the exchange call ringing cadence on the remote Communication Server user's telephone.

Redirection (Re-Ring)

This feature enlargement allows full control of the re-ring switchover timers in a DPNSS network for calls which are not answered by the called party. With this feature, the HiPath system can not only control the re-ring timers in the source node (as required in DPNSS networks), but also in the node from which the call was switched (as in CorNet networks).

Call Offer (Camp-On/Knocking)

This feature automatically applies the call waiting or knocking tone to a recipient's call when a further call is transferred to the user's line via an attendant (call offer). When the attendant transfers the call, the caller may either hear a ringing tone, or music on hold, or an announcement, depending on the system configuration. The busy, called party hears the knocking or call waiting tone. The party with whom the called user is connected does not hear the knocking tone.

Central Attendant Transfer (Night Service)

If an attendant console or attendant console group is not available, HiPath automatically transfers all calls to an alternative answering station (night service station).

This feature is implemented locally, and network-wide in the case of DPNSS and CorNet networks.

The following night service features are not implemented for DPNSS networks:

- Night service notification (message to satellite systems, that the attendant has activated the night service)
- Night service transfer via the same voice channel
- Automatic night service deactivation (used when attendant commences "day" service again).

10.4.9.5 Networked Name Displays (NNDIS)

This feature is used for calling and called line identification telephone displays of networked users throughout mixed DPNSS/CorNet network configurations. Names assigned to analog terminals are also displayed throughout the network.

10.4.9.6 Centralised Voice Mail (CVM)

This feature allows HiPath users to access and use HiPath voice mail equipment connected to a GPT system, provided this system is linked to the HiPath system via DPNSS/CorNet lines. In addition, GPT system users can access the voice mail service of a HiPath system, provided they are linked to the HiPath system via DPNSS/CorNet lines.

The feature allows network-wide call forwarding to the user's voice mailbox. "Message waiting indications" are also transmitted network-wide; the message is converted in the CDG.

The feature is only available for mixed system networks with HiPath and GPT DPNSS systems, and is only implemented for message exchange between these systems.

10.4.9.7 Do-Not-Disturb Override (DNDO)

The do-not-disturb feature enables users to send a "do-not-disturb" message (or busy signal) to callers.

The expanded feature with DNDO allows attendants and authorised users to override this state, provided the called party is not busy (i.e. is already engaged in a two-party connection in talking state).

10.5 Italy

10.5.1 Call Processing Features

10.5.1.1 End-of-Dial Detection for DID

In the case of DID from the analog exchange, the end-of-dialling signal is sent to the exchange

- after a fixed number of digits or
- after 4 to 8 s.

Both variants can be set by means of administration and maintenance.

In the first variant with a fixed number of digits, the waiting time may be limited. This time can be administered up to a value of 30 s since a timer is also started in the public exchange. After the timer has expired, busy tone is sent to the exchange but no end-of-dialling signal. The system waits for forward release.

10.5.1.2 Toll Exchange Forward Transfer Signal for Incoming Trunk Calls

After the internal user goes on-hook, the exchange does not release in this case but the toll exchange sends forward transfer signal. The forward transfer signal results in call signalling on the PABX AC.

10.5.1.3 Audible Tone Suppression for Incoming Calls without DID

In the case of incoming trunk calls via analog or digital trunks without DID, no audible tone is sent to the exchange before the internal user answers.

10.5.1.4 End-of-dial Signals with DID

In the case of incoming DID calls via analog trunks, the following is sent to the exchange after end-of-dialling:

- end-of-dial signal and
- ringback or busy tone.

In the case of digital DID trunks, the following is sent to the exchange after end-of-dialling:

- end-of-dial signal "User idle"
- end-of-dial signal "User busy"

  alternatively, via AMO only
- end-of-dial signal "User idle" with attendant intercept if user is busy.

Audible tones for the exchange are not required but have no negative effect either since the through-connection is only performed upon answering.

10.5.1.5 Attendant Override with DID

In the case of incoming DID via digital trunks to a busy user, the reception of the override signal results in

- attendant intercept (corresponds to "toll exchange override") (variant 1) or
- B-tone application to the exchange (variant 2).

If the internal user is busy in the case of DID from the exchange, the exchange receives the signal "End-of-dialling, user busy". The exchange subsequently applies B-tone. If the exchange then sends the override signal, the PABX switches the path through and also applies the B-tone. Thus, the calling external party hears B-tone from the exchange and then from the PABX.

10.5.1.6 Attendant Intercept with DID and Party Does Not Answer

The time after which a call is intercepted to the AC because the party does not answer can be set between 20 and 40 s.

10.5.1.7 Overriding an add-on conference

The feature "three-way conference intrusion" (override) is only implemented for the centralised attendant services, and not for standard users.

10.5.2 Analog trunk calls

Incoming, one-way, trunk calls with DID

Two-wire main station line for voice and signalling (DC, AC 25 Hz) (TMANI).

Signalling method:pulse dialling

Features:

- Idle state
- Seizure
- Dialling
- End-of-dialling
- Answering
- Backward release
- Long-distance exchange reringing with 25-Hz signal
- Attendant answering
- Exchange release

Notes on the criteria:

- Forward signals from the exchange by feeding in the normal (b-wire connected to positive conductor) or inverted state (a-wire connected to positive conductor) as well as loop interruption and change between high-level and low-level feeding or vice versa.
- Backward signals to the exchange by changing the loop closure from low-level to high-level and vice versa.
- Backward release after answering
- No idle state supervision in the PABX
- Call tracing by not going on-hook after end of call
- Toll exchange forward transfer signal before releasing exchange with attendant intercept any number of times (2 variants: 25 Hz manually from the toll AC or release with new seizure and selection)

Bothway Trunk Calls (Main Station Interface) with 50-Hz/12-kHz Call Data Registration

Two-wire main station line for voice and signalling (TMANI).

Signalling method:pulse dialling or DTMF

Line criteria for outgoing seizure:

- Idle state
- Seizure
- Acknowledgment
- Dialling
- Answering through evaluation of the 1st charge pulse
- Forward release
- Call charges
- New seizure disabled

Line criteria for incoming seizure

- Idle state
- Seizure
- Answering
- Forward release before answering
- Backward release after answering
- New seizure disabled

Notes:

1. Outgoing traffic

- 1A method for DTMF users and/or DTMF trunk
- After a seizure and before the start of dialling, trunk dial tone detection
- Special features of call charge pulse detection
  - Two consecutive call charge pulses with too short a pause in between are registered as a single call charge pulse.
  - Reception of a call charge pulse corresponds to answering.
  - The PABX must remain capable of recording call charge pulses throughout the entire reseizure lockout time.

2. Incoming traffic

- A seizure with 25-Hz ringing results in a call to the attendant console or to the night service
- Answering with a loop closure to the public exchange

10.5.3 Bothway Tie Traffic

Six-wire access line for voice (4-wire) and signalling (2-wire) on the basis of E&M or TF signalling (TMEMW, TMEW2 from HiPath 4000).

Signalling method:pulse dialling or DTMF

10.5.4 Digital Trunk Calls

10.5.4.1 Incoming, One-Way Trunk Traffic with/without DID

Four-wire main station line for voice with channel-associated signalling (DIU-CAS).

Signalling method:pulse dialling

1. Criteria for incoming calls with DID:
   - Idle state
   - Seizure signal 1 (normal telephone seizure)
   - Seizure signal 2 (64-Kbit/s channel)
   - Seizure signal 3 (test seizure)
   - Ready condition of the PABX
   - Pulse dialling
   - End-of-dialling idle to the exchange
   - End-of-dialling busy to the exchange
   - Answering to the exchange
   - Long-distance exchange callback after backward release
   - Forward release to the exchange
   - Release check from the PABX
   - Idle from the PABX after blocking
   - Backward release from the PABX after answering
   - Answering from the attendant console
   - Offering on end-of-dial busy (override for user or attendant intercept or ignore)
   - Release from the exchange before answering

- Barring from the PABX
- Unbarring from the PABX
- Ready state from PABX

2. Criteria for incoming calls without DID:
   - Idle state
   - Seizure signal 1
   - Seizure signal 2
   - Seizure signal 3
   - Answering to the exchange
   - Long-distance exchange callback after backward release
   - Barring from the PABX
   - Forward release from the exchange
   - Release check from the PABX
   - Idle from the PABX after blocking
   - Ready state from the PABX
   - Backward release from the PABX after answering
3. Special criteria:
   - Pulse dialling
   - Pulse dialling times

Pulse:	42 - 58 ms
Pulse plus pause:	90 - 100 ms
Interdigit pause:	≥500 ms

   - Only waiting without answering for reception of forward release
   - If end-of-dialling = free, the attendant intercepts after 20 - 40 s if the user does not answer.
   - If dialling is incomplete, the attendant intercepts after 4 - 8 s.
   - Is possible to administer two end-of-dialling alternatives:
     – End-of-dialling free/busy signal, depending on the user status
     – End-of-dialling free signal and attendant intercept whenever the user is busy.

Since the call is not switched in the exchange until the PABX answers, it is not essential to transmit any audible tones, though on the other hand it does not do any harm.

- No maximum times are defined for callback and offer pulses. A system message must be output after the minimum time.
- Minimum detection time for offer after the end-of-dialling: 100 ms
- Any number of offer pulses can be transmitted.
- An offer from the exchange leads in terms of administration to:
  - An attendant intercept
  - Transmission of the busy tone up to the forward release
- Long-distance exchange reringing is possible any number of times.
- Call tracing is activated by the called user not going on-hook at the end of the call until the exchange console has answered. There is no special line signal to or from the exchange.
- The release check is transmitted at the latest 400 ms after the forward release has been detected.

4. Alarm display of faulted lines (PCM carrier alarm) after 4 - 6 min.
5. Restrictions:
   - only standard seizure (with other types of seizure: wait for forward release)
   - the "offer" signal results in attendant intercept
   - call tracing is initiated by "flash hook" by the user

10.5.4.2 Outgoing, One-Way Trunk Calls without DID

Four-wire main station line for voice with channel-associated signalling (DIU-CAS).

Signalling method:pulse dialling or DTMF

1. Criteria for outgoing calls:
   - Idle state
   - Seizure signal 1
   - Seizure signal 2
   - Ready condition from the exchange
   - Dialling (pulse dialling, DTMF)

- End-of-dialling signal from the exchange (not ensured)
- Answering from the exchange
- Register callback from the PABX with DTMF dialling
- Metering pulses from the exchange
- Forward release from the PABX
- Release check from the exchange
- Idle state from the exchange after barring
- Barring from the exchange
- Barring from the exchange if release check signal does not arrive within 30 s
- Forced release from the exchange (backward release)

2. Special criteria:
   - Pulse dialling or DTMF
   - Pulse dialling times

Pulse:	42-58 ms
Pulse plus pause:	90-100 ms
Interdigit pause:	≥700 ms

- DTMF dialling/pause ratio: 70/70 ms
- Only exchange barring possible.
- The answer signal always precedes the first call charge pulse.
- Answering is possible immediately following the end-of-dialling (change from 10 to 00).
- If no release control signal is received within 30 s in the case of forward release, 11 is transmitted until 11 is received. At the same time, a timer of 4 - 6 min. is started; when this timer expires, an alarm is output. If 11 is received, reset to 01 is performed. The transmission is then available again if 01 is received.
- If the release control signal is >3000 ms, this is barring from the exchange. A timer of 4 - 6 min. is started; when this timer expires, an alarm is output.
- Release control signals <60 ms are ignored.
- Forced release from the exchange will be recognised in each condition.
- No dial tone detection. There should not be a dialling output until 700 ms after the ready condition has been detected.

- Minimum time between the forward release and the seizure: 70 - 100 ms.
- There is no end-of-dialling signal from the exchange.
- The number-received signal from the exchange causes the dialling output to stop immediately.
- The end-of-dialling pulse is only checked with regard to its minimum length.
- Call charge pulses are only supervised with regard to the minimum pulse length and pulse pauses.
- In the talking state, failures on the a-, b-bits do not result in release.
- If no seizure acknowledgment is detected for an outgoing seizure after a fixed period, the system waits for a seizure acknowledgment for an unlimited period of time and releases the connection when one is received.
- The exchange does not answer a second time.

3. Alarm display of faulted lines (PCM carrier alarm) after 4 - 6 min.
4. Restrictions:
   - only standard seizure
   - no register callback
   - no special services
   - no suffix dialling after end-of-dialling

10.5.5 Main/Satellite PABX Traffic with Bothway Dc Loop Signalling (TMBS, TMCL)

Main/satellite PABX traffic is handled using bothway dc loop signalling via modules TMBS and TMCL.

Impedance matching is carried out by modifying the existing transmission equipment used in the Fed. Rep. of Germany.

10.5.6 Audible Tones for Italy

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
1st exchange dial tone	425	<u>**200**</u>/200/<u>**600**</u>/1000
2nd exchange dial tone		

Tone	Frequency [Hz]	Pulse / Pause [ms]
Busy tone	425	**500**/500
ATB tone	425	**200**/200

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse/Pause [ms]	Level (dBm0)
Internal dial tone	1	350+425	Continuous tone	-6
External dial tone	2	425	**200**/200/**600**/1000	-6
Ringback tone	4	425	**1000**/4000	-6
Busy tone	5	425	**500**/500	-6
Override tone	6	425	**200**/200/**200**/1400	-11
Call waiting tone	7	425	**100**/4900	-11
Special dial tone	3	425	Continuous tone	-6
NU				
Data call tone				
Positive acknowledgement tone	10	425	**100**/100/**100**/100/**100**/1500	-6
ATB tone	11	425	**200**/200	-6
Conference tone	12	425	**200**/9800	-11
Abandonment request tone	23	425	**100**/100	-6
Conference auxiliary tone	24	425	Continuous tone	-11
LCR	26	1800	**340**/200/**340**/200/**340**/1000	-6

Special tone sequences:

Tone	Sequence	derived from
Conference tone	200 ms Continuous	Conference auxiliary tone Conference tone
Call waiting tone	Continuous	Call waiting tone
Call waiting tone for night service terminal	Continuous	Call waiting tone
Override tone	Continuous	Override tone

10.5.7 PCM30 Characteristics

2-bit signalling in 16th time slot

- Send direction:c=0, d=1
  Receive direction:c, d are ignored

Frame alignment signals / answering codes

- Time slot 0
  0th, 2nd, ..., 30th frame
  Frame alignment signal: X0011011 with X = 1
- 1st, 3rd, ..., 31st frame
  Pulse frame answering code: X1DNYYYY with X = Y = N = 1
  D = urgent alarm bit 3 (no alarm = 0)
- 16th time slot / 2nd nibble, 0th frame
  Superframe answering code: ydny with y = n = 1
  d = urgent alarm bit 6 (no alarm = 0)

Alarm display of faulted lines (PCM carrier alarm) after 4 - 6 min.

Channel-individual display of alarms and barrings

Idle code on the speech channels 54H

10.5.8 Transmission Requirements

Levels and impedances have been adapted in accordance with Section 12 of the "Norme Techniche per Centralini Numerici (ISPT)":

12.1.2	for analog lines
12.1.3	for analog 2-wire transmission
12.1.12	for analog 4-wire tie lines

10.5.9 Error and Barring Displays for PCM 30 Channels

Channel-individual display of alarms and barrings.

10.6 Luxembourg

10.6.1 Analog Trunk Calls

Bothway Trunk Calls with DID for F Signals and Pulse Dialling (TMFS)

Two-wire main station line for voice and signalling.

Signalling method:pulse dialling

Features:

- Ready for operation (line supervision)
- I/c seizure
- I/c dialling
- End-of-dialling
- Remote override
- Answering
- Forward release
- Backward release
- Short-to-ground monitoring
- O/g seizure
- O/g dialling
- Release
- 16-kHz charge reception

Bothway Main Station Line with Pulse Dialling or DTMF Dialling (TMANI)

Two-wire main station line to the central office for voice and signalling.

The line can be operated on a

- bothway
- o/g only
- i/c only

basis.

Signalling method:pulse dialling or DTMF

I/c traffic always results in attendant intercept.

Features:

- Incoming traffic
  - Seizure (ringing ac)
  - Answering
  - Forward release
  - Backward release
- Outgoing traffic
  - Seizure

  - Dialling
  - Forward release
  - 50-Hz/16-kHz charge reception

10.6.2 Digital Exchange Traffic

Incoming Traffic with DID and MFC-R2 (DIU-CAS)

Signalling method:MFC-R2

Features:

- Initial seizure
- Start-of-dialling (from PABX)
- Dialling information signalling
- Answering (from PABX)
- Forward release
- Backward release
- Blocking (from PABX)
- Register recall

Outgoing Traffic with DTMF Signalling (DIU-CAS)

Signalling method:DTMF

Features:

- Initial seizure
- Start-of-dialling (from exchange)
- Dialling information signalling
- Answering (from exchange)
- Call charge pulse from exchange
- Forward release
- Backward release
- Blocking (from exchange)
- Register recall

10.7 Finland

10.7.1 Call Processing Features with MFC-R2 Signalling

10.7.1.1 Override in the event of Busy Station

In the case of manually switched trunk calls the attendant in the long-distance exchange has the option of allowing the call to be intercepted by the PABX attendant console if a PABX station is busy.

Attendant intercept takes place as soon as the PABX has received the override signal from the exchange.

10.7.1.2 A-user Number Identification for Outgoing and Incoming Trunk Calls

In incoming and outgoing trunk calls, A-user number identification is effective while the call is being set up.

The following two variants are employed for identification
:

Variant 1:	Identification by applying the backward signal A9
Variant 2:	Identification by twice applying the backward signal A5 (after interrogation of the B-user number)

Both variants are to be set up separately for incoming and outgoing trunk calls, controllable by parameters (AMO).

Variant 1 is used in the case of MFC-R2 connections to more modern exchanges, Variant 2 in the case of MFC-R2 connections to older exchanges.

A-user Number Identification for Outgoing Trunk Calls

With both variants, A-user number identification proceeds from the exchange. If the PABX is unable to carry out A-user number identification (e.g. if in the case of main/satellite PABX traffic with the A-user in the satellite PABX the transmission system between main and satellite system is not MFC-R2), the PABX rejects the identification request (with the rejection signal "I12").

With Variant 1 the exchange acknowledges any Group I forward signal with the backward signal "A9", thereby requesting the A-user number from the PABX. The PABX responds by transferring the A-user number. The exchange acknowledges each transmitted A-user number signal with the signal "A9".

The PABX is able to process the signal "A9" in pulse form (in the absence of a forward signal). As an essential requirement, the complete B-user number must be transmitted first.

With Variant 2 the exchange only requests the A-user number, by twice applying the backward signal "A5", after the B-user number has been completely interrogated.

The PABX responds to the request by transferring the A-user number. The exchange acknowledges each transmitted A-user number signal with "A5".

Regardless of the two variants 1 or 2, the A-user number in the PABX consists

- either of the three signal groups
  - Discriminating digit "9"
  - Network group number (discriminating digit + network group number = 2 - 4 digits)
  - Extension DID number (max. 9 digits)
- or only of the
  - Extension user number (max. 5 digits).

Either possibility can be set up as an alternative.

The individual signal groups are transmitted to the exchange without any separators.

A-user Number Identification for Incoming Trunk Calls

With both variants, A-user number identification proceeds from the PABX. If the exchange is unable to carry out A-user number identification (e.g. because the exchange is not equipped for it), the exchange rejects the identification request (with the rejection signal "I12").

Sequence of functions for variant 1 or 2 analogous to outgoing trunk calls.

The A-user number transmitted by the exchange consists

- In the case of 1 of the three signal groups
  - Discriminating digit "9"
  - Network group number (discriminating digit + network group number = 2 - 4 digits)
  - Exchange party's call number (max. 9 digits)
- in the case of 2 only of the
  - Exchange party's call number (max. 9 digits).

The exchange party can also be a PABX user.

Either possibility can be set up as an alternative.

The individual signal groups are transmitted to the PABX without any separators and are only stored in the PABX as overall information (and, where applicable, fed out on a service terminal).

10.7.1.3 Call Tracing for Incoming and Outgoing Trunk Calls with MFC-R2 Signalling

This feature can only be activated for incoming and outgoing trunk calls during the call phase.

Restrictions:

- Call tracing is not possible in the HiPath network (the user's call tracing classmark cannot be transmitted throughout the network).
- If the HiPath system carries out a soft/hard restart, lines (circuits) held for call tracing are released.
- The max. number of trunks simultaneously held for call tracing can be set and changed by means of an AMO. Each time a call tracing request is made, the PABX checks whether this number has been reached. If it has, the internal user's order is negatively acknowledged.

Call Tracing for Incoming Trunk Calls

This feature can be used by the following PABX users (B-users):

- Attendant console
- Night answer extension
- PABX extension user.

Call tracing orders are automatically registered in the PABX at the service terminal. The following data is registered:

- Date and time of the start of call tracing
- Call number of the PABX user
- Number of the trunk (hardware PEN)
- Exchange party number (if applicable)
- Additional text information (if applicable).

Call tracing classmark:

- It is an essential condition of using the call tracing feature that the PABX user has the call tracing classmark (can be set up with a MML command; no privileged code required for EMML).

- As soon as **one** PABX extension user is given the call tracing classmark, all attendant consoles and night answer extensions must automatically have this classmark as well.
- If individual trunks have been barred for call tracing by the PABX it prevents the feature from being activated.

In the case of MFC-R2 signalling, the calling exchange subscriber can be identified or traced in the following two ways:

- By identifying the number of the A-user
- By tracing calls when the B-user is free if A-user number identification cannot be carried out in the exchange.

A-user number identification

As a prerequisite for this type of identification, the feature "A-user identification for incoming trunk calls" must have been set up in the PABX.

The calling outside party's number is already ascertained via the A-user number identification during the call setup and buffered in the PABX for the entire call duration. When the PABX user activates call tracing, this number is logged in the call tracing data, so additional holding of the trunk is not necessary.

In the case of attendant intercept, call transfer/pickup or call forwarding to a PABX user with the call tracing classmark, implementation of a call tracing order is permitted in the PABX in the subsequent call status.

The call is terminated in the usual way when the outside or PABX user goes on-hook.

Tracing calls when the B-user is free

This type of identification is to be employed whenever A-user number identification has not been set up in PABX or has not been successfully implemented.

In the case of an incoming call to a PABX user with the call tracing classmark, at the end of the call setup the PABX sends the MFC signal B1 for call tracing to the exchange. This prevents the exchange from forward releasing the trunk. The release initiative comes from the PABX by means of a clearing signal.

In the case of attendant intercept, call transfer/pickup or call forwarding to a PABX user with the call tracing classmark, implementation of a call tracing order is permitted in the PABX in the subsequent call status. The identification for call tracing to the exchange remains in force. If the new PABX user does not have the call tracing classmark, call tracing to the exchange remains in force.

If a PABX user activates call tracing in the call status and this call tracing order is accepted by the PABX, the call tracing data are automatically registered on the service terminal.

The call tracing order is **not** signalled to the exchange.

If the PABX user goes on-hook after activating call tracing, the internal connection is released by the PABX. The trunk is put in the call tracing status. A trunk "held" in this way can only be released by the exchange or by an administration and maintenance order from the PABX.

After activation of call tracing, the call can be transferred to or picked up by a new PABX user. If the new PABX user has the call tracing classmark, the PABX allows another call tracing order to be implemented in the subsequent call status.

If the PABX user goes on-hook without having activated call tracing, the connection is released in the usual way in the backward direction.

Sequence of operations in the call tracing procedure

The call tracing procedure can only be activated by the internal user during the call phase in the following manner:

For analog terminals

Initiate consultation by pressing the signal key (flash or ground key) and dial the call tracing code. The call tracing code can be set up on a system-specific basis for all PABX users (number of digits: 2 or 3). After initiation of consultation the trunk is held by the PABX. During the holding procedure the waiting status is signalled to the external user by music or an announcement.

Acknowledgments

If call tracing is accepted by the PABX, "positive acknowledgment tone" is applied.
Duration: generally 3.5 s, can be set by administration and maintenance.

If call tracing is not accepted by the PABX, "negative acknowledgment tone" is applied.
Duration: generally 3.5 s, can be set by administration and maintenance.

After the acknowledgment tone time the connection is automatically switched through to the trunk.

For digital terminals

Dial the call tracing code either during the call or by pressing the repertory/destination key set up for it; at the attendant console only by pressing the call tracing key. The connection to the outside party is not interrupted, the internal party remains in the call status.

Acknowledgments

- for digital terminals (except for attendant console), on the display as follows:
  - Text display "CALLER TRACED" in the local language, if call tracing is accepted by the PABX.

  - Text display "NOT AUTHORISED", "NOT POSSIBLE" or "PLEASE REPEAT" in the local language, if call tracing is not accepted by the PABX.

- for the attendant console on the screen as follows:
  - Text display "STORED" in the local language, if call tracing was accepted by the PABX.
  - Text display "NOT POSSIBLE" or "PLEASE REPEAT" in the local language, if call tracing is not accepted by the PABX.

Printout of call tracing data

A call tracing order accepted by the PABX is immediately registered on the service terminal with the following parameters

- for "A-user number identification"
  - Date and time at start of call tracing
  - Call number of the PABX user
  - Number of the trunk (hardware PEN)
  - A-user number transmitted from the exchange
- for "Tracing calls when the B-user is free"
  - Date and time at start of call tracing
  - Call number of the PABX user
  - Number of the trunk (hardware PEN)

If the exchange releases a held trunk connection in the case of the identification type "Tracing calls when the B-user is free" within a specified period (e.g. 60 s) after the PABX user has gone on-hook, a non-executed call tracing order in the exchange is assumed.

This is registered in the PABX by an immediate second printout at the service terminal with the following parameters:

- Date and time of the release of the held call by the exchange
- Call number of the PABX user (as an allocation point for the 1st call tracing printout)
- Supplementary text "RELEASE DESPITE CALL TRACING" (in English).

Call Tracing for Outgoing Trunk Calls

As already mentioned under "Call tracing for incoming trunk calls", with MFC-R2 signalling the calling PABX user can also be identified or traced in the following two ways for outgoing trunk calls:

- through A-user number identification
- through call tracing when the B-user is free, if A-user number identification cannot be implemented.

A-user number identification

With A-user number identification the exchange already requests the A-user number of the PABX user during call setup and buffers it in the exchange. The exchange is thus able to identify the initiator if required.

The call is terminated normally if the exchange or PABX user goes on-hook.

Call tracing if the B-user is free

In the case of an outgoing trunk call, the exchange reports by means of the MFC signal B1 at the end of the connection setup that an outside party with the call tracing classmark has been called. This prevents the exchange from forward releasing. The release initiative comes from the exchange.

If the outside user activates call tracing during the call phase, the PABX is not informed of the call tracing order by the exchange.

If the exchange user goes on-hook without having activated call tracing, the PABX receives the clearing signal from the exchange and the connection is released in the usual way in the forward direction.

If the exchange does not release an outgoing trunk connection within a specified period (e.g. 60 s) after the PABX user has gone on-hook, a held trunk is assumed. This is immediately registered in the PABX with a printout of the call tracing data on the service terminal with the following parameters:

- Date and time at start of call tracing
- Call number of the PABX user
- Number of the trunk (hardware location)
- A-user number transmitted to the exchange.

A held trunk can only be released by the exchange or by an administration and maintenance order from the PABX.

Call charge pulses are registered in the call tracing status.

10.7.1.4 Call Forwarding (all calls) to the Exchange

This feature can only be employed by the PABX extension users (excluding the night answer extension) to whom a separate classmark for call forwarding (all calls) to the exchange has been assigned (can be set up using MML command; no privileged ID required for EMML).

The attendant console has no classmark for call forwarding (all calls) to the exchange.

Call forwarding (all calls) to the exchange can only be assigned to one user at a time (call forwarding cannot be chained).

The destination user, to whom a call is forwarded, may also be a PABX user (and thus a HiPath user as well).

Call forwarding (all calls) to the exchange is initiated and activated for Anate and Digite telephones using the procedures currently employed in HiPath.

If an incoming call is forwarded back to the exchange by the PABX, this call is set up as a separate connection. A connection is switched from the calling exchange party to the forwarded user in the PABX after the forwarded user has answered.

Any call charges for the forwarded call are debited to the PABX extension user who activated call forwarding.

10.7.1.5 Call Processing Features for N2 Signalling

Call Tracing for Incoming and Outgoing Trunk Calls

This feature can only be activated for incoming and outgoing trunk calls during the call phase.

Restrictions:

- Call tracing is not possible in the HiPath network (the call tracing classmark of the user cannot be transferred throughout the network).
- If the HiPath system effects a hard or soft restart, any lines (circuits) which have been traced are released.
- The maximum number of lines which can be traced at any one time can be set and altered by means of an administration & maintenance order.
- It is possible to buffer the call tracing data with the intention of being able to retrieve it again at a later date if all the data relevant to operation is stored on a suitable medium, e.g. on hard disk. It is not possible to retrieve call tracing data which has been stored on this type of medium selectively.

Call Tracing for Incoming Trunk Calls

The feature can be used by the following PABX users (B-users):

- Attendant console
- Night answer extension
- PABX extension user.

Call tracing orders are registered automatically in the PABX on the service terminal.

The following data is registered:

- Date and time of the start of call tracing
- Call number of the B-user
- Number of the trunk
- Additional text information (if applicable).

Call tracing classmark

- It is an essential condition of using the call tracing feature that the PABX user has the call tracing classmark (to be set up by means on an MML command; no privileged ID for EMML necessary).
- As soon as **one** PABX extension user is given the call tracing classmark, all attendant consoles and night answer extensions automatically have this classmark as well.
- If individual trunks have been barred for call tracing by the PABX, the latter prevents the feature from being activated.**Call tracing criterion in the end-of-dialling state**

If the PABX user has the call tracing classmark, the PABX sends the "Call tracing" signal to the exchange in the end-of-dialling state. This prevents forward release in the exchange. The release initiative comes exclusively from the PABX in the form of a clearing signal.
If the PABX user with the call tracing classmark does not answer after the end-of-dialling within 30 - 50 s (variably settable), and there is no call forwarding (no answer) or attendant intercept, the exchange is informed of this by cancelling of the "Call tracing" signal.

If the PABX user does not have the call tracing classmark, the exchange receives the "end-of-dialling" signal. In this case the PABX does not start a watchdog (30 - 50 s) if the PABX user does not answer.

If attendant intercept or call forwarding (no answer) has been set, the PABX checks the call tracing classmark of the new PABX user again, and informs the exchange if it has one.

If the PABX user is busy, the call tracing classmark is irrelevant.

Call tracing criterion in the answer state

For a PABX user with the call tracing classmark, the signal "Answer call tracing" is transmitted to the exchange in the answer state.
For a PABX user who does not have the call tracing classmark, the exchange receives the "Answer" signal.

If a call is transferred/picked up, the exchange is informed of the new PABX user's call tracing classmark on answering.

If the PABX user goes on-hook without having activated the call tracing feature, the exchange is notified by means of a clearing signal. It then releases the connection to the PABX.

Sequence of operations in the user call tracing procedure

"Call tracing" can only be activated in the call status as follows by the internal user:

For analog terminals

Initiate consultation by pressing the signal key (flash or ground key) and dial the call tracing code. The call tracing code can be set up on a system-specific basis for all PABX users (number of digits: 2 or 3). After initiation of consultation the trunk is held by the PABX. During the holding procedure the waiting status is signalled to the external user by music or an announcement.

Acknowledgments

- If call tracing is accepted by the PABX, "positive acknowledgment tone" is applied.
  Duration: generally 3.5 s, can be set by administration and maintenance.
- If call tracing is not accepted by the PABX, "negative acknowledgment tone" is applied.
  Duration: generally 3.5 s, can be set by administration and maintenance.

After the acknowledgment tone time the connection is automatically switched through to the trunk.

For digital terminals

Dial the call tracing code either during the call or by pressing the repertory/ destination key set up for it; at the attendant console only by pressing the call tracing key. The connection to the outside party is not interrupted, the internal party remains in the call status.

Acknowledgments

- for digital terminals (except for attendant console), on the display:
  – Text display "CALLER TRACED" in the local language, if call tracing is accepted by the PABX.
  – Text display "NOT AUTHORISED", "NOT POSSIBLE" or "PLEASE REPEAT" in the local language, if call tracing is not accepted by the PABX.
- for the attendant console on the screen as follows:

- Text display "STORED" in the local language, if call tracing was accepted by the PABX.
- Text display "NOT POSSIBLE" or "PLEASE REPEAT" in the local language, if call tracing is not accepted by the PABX.

The exchange is not informed by the PABX about the call tracing order at this point (i.e. the "Answer call tracing" signal is maintained). The PABX does not send the "Fixed call tracing" signal to the exchange, and release the connection internally, until the PABX user goes on-hook. The PABX/exchange connection is maintained in the call tracing state.

If the connection is transferred to or from a new PABX user after "call tracing" has been initiated, the "Answer call tracing" state remains active, even if the new PABX user does not have the call tracing classmark. The "Fixed call tracing" signal is transmitted to the exchange when the new PABX user goes on-hook.

A traced connection can only be put in the idle condition by means of an administration and maintenance order from the PABX or the exchange.

Printout of call tracing data

A call tracing order accepted by the PABX is immediately registered on the service terminal.

If, however, the exchange releases a held trunk connection within a specified period (e.g. 60 s) after the PABX user has gone on-hook, a non-executed call tracing order in the exchange is assumed. This is immediately registered in the PABX by a second printout at the service terminal with the following parameters:

- Date and time of the release of the held call by the exchange
- Call number of the PABX user (as an allocation point for the 1st call tracing printout)
- Supplementary text "RELEASE DESPITE CALL TRACING" (in English)

If the connection is transferred to or from a new B-user after "call tracing" has been initiated, the "Answer call tracing" state remains active, even if the new B-user does not have the call tracing classmark. The "Call tracing" line signal is transmitted to the exchange when the new B-user goes on-hook.

If "call tracing" is accepted by the exchange, the latter causes the exchange/ PABX connection to be maintained. The connection can only be made to assume the idle state by an administration and maintenance order from the PABX or the exchange.

If, however, "call tracing" is **not** accepted by the exchange, the PABX is informed when the external user goes on-hook by means of a "Forward release" signal. In this case the trunk call and the internal call are released (if the B-user has not yet gone on-hook), and the circuit (channel) is made to assume the idle state. The additional text information "Exchange released despite call tracing" is output as part of the call tracing data which must be registered on the service terminal.

The PABX checks on reception of the "Forward release" line signal whether or not the B-user has previously issued a call tracing order. If so, it has not been accepted by the exchange.

Call Tracing for Outgoing Trunk Calls

During outgoing trunk calls the PABX can receive the call tracing criterion from the exchange at the following points:

- in the end-of-dialling state (by means of the "Call tracing" signal)
- in the answer state (by means of the "Answer call tracing" signal)
- in the end-of-call state (by means of the "Fixed call tracing" signal).

These signals inform the PABX that the trunk call must not be released from the PABX after the A-user goes on-hook. The release initiative comes from the exchange or in the form of an administration & maintenance order from the PABX.

The minimum time for detecting the call tracing criteria in the PABX is 300 ms.

In the event of interrupts >400ms after a call tracing criterion has been detected, the call tracing state is cancelled for the A-user in the PABX.

Charge pulses in the call tracing state are to be registered

If the exchange user activates the "Call tracing" feature during the call phase, the PABX is only informed of the call tracing order, by means of "Fixed call tracing" signalling, after the exchange user has gone on-hook (i.e. the original "Answer call tracing" signal remains on the line until the the B-user goes on-hook, and then changes to "Fixed call tracing").

Each call tracing order is registered in the PABX at the service terminal with the following data:

- Date and time of detection of the call tracing signal "Fixed call tracing"
- Call number of the PABX user
- Number of the trunk (hardware PEN)
- Digits transmitted to the exchange.

If the exchange user goes on-hook without having activated call tracing, the exchange sends the clearing signal to the PABX; the PABX reacts by releasing the connection in the usual way in the forward direction.

If the PABX user is the first to go on-hook without the PABX having received a call tracing criterion ("Call tracing", "Answer call tracing", "Fixed call tracing"), the connection is released in the usual way in the forward direction.

This also applies if there was a call tracing criterion on the line at an earlier time, but one which was removed. In this case there is no registration of call tracing data.

If the PABX user is the first to go on-hook and if the exchange user previously activated the "Call tracing" feature, the call tracing data are registered on the service terminal after the exchange user has gone on-hook.

If the exchange user does not activate the "Call tracing" features until the PABX user has gone on-hook, the PABX receives the "Fixed call tracing" signal from the exchange after the exchange user goes on-hook. This also results in registration of the call tracing data on the service terminal.

A held trunk call is released by the PABX:

- after receipt of the clearing signal or idle state code from the exchange,
- by means of an administration and maintenance order from the service terminal of the PABX.

10.7.2 Analog Trunk Calls

Incoming Trunk Calls with DID for Pulse Dialling with N2 Signalling (TMN2S)

Three-wire main station line for voice and signalling.

Signalling method:pulse dialling

Features:

- Seizure
- Dialling
- Answering
- Forward release
- Backward release
- Short-to-ground monitoring

Bothway Main Station Line with Pulse Dialling without DID (TMANI)

Two-wire main station line to the central office for voice and signalling.

The line can be operated on a

- bothway
- o/g only
- i/c only

basis.

Signalling method:pulse dialling

Incoming traffic always results in attendant intercept.

Features:

- Incoming traffic
  - Seizure (ringing ac)
  - Answering
  - Forward release
  - Backward release
- Outgoing traffic
  - Seizure
  - Dialling
  - Forward release
  - 16-kHz charge reception

10.7.3 Digital Trunk Calls

Only one-way operation (either incoming or outgoing) is possible in conjunction with N2 signalling (incoming only with direct inward dialling).

The Helsinki Telephony Company (HTC) bit protocol type 3 is used.

The following feature is not implemented:

- Transparent data transmission at 64 kbit/s via the public DIGINET network

10.7.3.1 Incoming Call with DID and N2 Signalling for Pulse Dialling

Signalling method: pulse dialling

Features:

- Seizure
- Start-of-pulsing
- End-of-dialling
- Busy
- Answering
- Clearing signal
- Forward release

- Forward release acknowledgment
- Blocking
- Remote override

10.7.3.2 Outgoing Call for Pulse Dialling with N2 Signalling

Signalling method: pulse dialling

Features:

- Seizure
- Start-of-pulsing
- End-of-dialling
- Answering
- Charge meter pulses
- Backward release
- Forward release
- Forward release acknowledgment
- Blocking
- Reanswering in outgoing trunk calls

10.7.3.3 Bothway Trunk Calls with/without Direct Inward Dialling (MFC-R2 as per Q.421)

The bits in signalling channel 16 of the PCM30 connection are used for line signalling (as per ITU G.732). Only bits a and b are used in both directions; bits c and d are fixed to c=0 and d=1 (DIU-CAS).

The line signals are sent either as a continuous signal or as a pulse and, with the exception of charge pulse, release request, and override, are identical in both directions.

Bothway traffic has been implemented.

For incoming and outgoing calls the link between the PABX and the exchange forms a separate MFC-R2 signalling section. The exchange employs an intermediate register signalling to the PABX.

Signalling method:MFC-R2

Features:

- Idle state
- Seizure
- Seizure acknowledgment
- Answering
- Call charge pulse (outgoing traffic)
- Clearing signal
- Release request (outgoing traffic)
- Release
- Backward blocking
- Release acknowledgment
- Override (busy PABX user, incoming traffic)
- Release acknowledgment
  - Duration of charge pulse:150 ms
  - Release request:Continuous signal or 600-ms pulse
  - Override:
    as a continuous signal on bit b; attendant intercept does not take place until the signal has been applied for ≥700 ms and has thus been reliably detected.
    as a pulse on bit a; start and end of override is received as a 150-ms pulse on the a-bit.

10.7.4 Audible Tones for Finland

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone	425	Continuous tone
Busy tone	425	**300**/300
ATB tone	425	**300**/300
Internal dial tone	425	**200**/300/**200**/300/**200**/800

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse/Pause [ms]	Level (dBm0)
Internal dial tone	1	425	**200**/300/**200**/300/**200**/800	-5.5
External dial tone	2	425	Continuous tone	-5.5

Tone	Tsl	Frequency [Hz]	**Pulse**/Pause [ms]	Level (dBm0)
Ringback tone	4	425	**1000**/4000	-2.5
Busy tone	5	425	**300**/300	-2.5
Override tone	6	425	**200**/300/**200**/1300	-12.5
Call waiting tone				
Special dial tone	3	425	**600**/25 (**660**/20)	-5.5
NU tone	8	950/1400/ 1800	Triple tone **332-332-332**/1000	-2.5
Data call tone	9	1300	**600**/1800	-1.5
Conference auxiliary tone	24	425	Continuous tone	-12.5
LCR	26	1800	**340**/200/**340**/200/**340**/1000	-2.5

Special tone sequences:

Tone	Sequence	derived from
Conference tone	200 ms 300 ms 200 ms 5000 ms 200 ms 300 ms 200 ms Continuous	Conference auxiliary tone Pause Conference auxiliary tone Pause Conference auxiliary tone Pause Conference auxiliary tone Silence
Call waiting tone		
Call waiting tone for night service terminal		
Override tone	Continuous	Override tone

10.8 Switzerland

10.8.1 Analog Trunk Calls

Incoming Trunk Calls with DID for Pulse Dialling and MFC-R2 Signalling with Loop-In Signal Loop (TMANI)

Two-wire loop-in signal loop access line for voice and signalling.

Signalling method:pulse dialling, MFC-R2

Loop signalling with feeding from the public exchange is the basic signalling. Feeding from the PABX only takes place in the disabled state.

SKZ mode

Features:

- Seizure
- Dialling
- Answering
- Forward release
- Backward release
- Barring

Bothway Main Station Line with Pulse Dialling and DTMF (TMACH)

Two-wire main station line to the central office for voice and signalling.

The line can be operated

- bothway,
- in outgoing direction only,
- in incoming direction only.

Signalling method: pulse dialling, DTMF

Features:

- Incoming calls
  - Seizure (AC ringing voltage)
  - Answering
  - Forward release possible if exchange set up correspondingly
  - Backward release
- Outgoing calls
  - Seizure
  - Dialling
  - Forward release
  - 12-kHz call charge reception

Grounding of the operating loop for consultation hold and transfer for tie traffic use with trunks.

10.8.2 Tie traffic

Bothway Tie Connection for Pulse Dialling with Status or Start-Stop Signalling.

Eight-wire bothway tie line with

- 4 wires for voice paths
- 4 wire for signalling (with 2-wire or 4-wire signal wire, depending on interface used).

Signalling method:pulse dialling, DTMF

10.8.3 Bothway ISDN Trunk Circuit with SWISSNET 2 (S_0 and S_2)

SWISSNET 2 (SN2) is "ISDN subscriber line circuit and trunk circuit in the public network" in Switzerland. SN2 is based on the ITU recommendation issue Blue Book and the corresponding CEPT recommendation.

Features

- Basic services
  - Connection setup and cleardown for voice (A-law)
  - TD 64 64 kbit/s transmission service
  - TELEFAX 3 3.1 kHz audio
  - TELEFAX 4 64 kbit/s
  - TELETEX 64 kbit/s
  - Audio 3.1 kHz transmission service
- Additional services
  - DID in subscriber PABX
  - Display, suppress, force and register identification
  - Subaddressing
  - Call charge information for the subscriber

  Features not implemented
  - Unconditioned call forwarding
  - Call forwarding to standard voice text
  - Closed user group

– Predefined connection

10.8.4 Audible Tones for Switzerland

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone	425	Continuous tone
Busy tone	425	**250**/250
ATB tone	425	**200**/200
Internal dial tone	425	Continuous tone

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse / Pause [ms]	Level (dBm0)
Ringing verification tone	4	425	**1000**/4000	-6
Busy tone	5	425	**500**/500	-6
Override tone	6	1400	**200**/2000	-13.5
Call waiting tone	7	425	**200**/200/**200**/4000	-13.5
Data call tone	9	1300	**600**/1800	-1
ATB tone	11	425	**200**/200	-6
Spezialsummton	3	425/425+340	**1100**/1100	-2.9
Internal buzzer tone	1	425	Continuous tone	-1
External buzzer tone	2			
NU tone	8	950/1400/ 1800	Triple tone **340-340-340**/1000	-6
Conference auxiliary tone 2	23	425	Continuous tone	-13.5
Conference auxiliary tone 2	24	1400	Continuous tone	-13.5
LCR	26	1800	**340**/200/**340**/200/**340**/1000	- 6

Special tone sequences:

Tone	Sequence	derived from
Conference tone	200 ms 2000 ms 200 ms Continuous	Conference auxiliary tone Pause Conference auxiliary tone Silence
Call waiting tone	Continuous	Call waiting tone
Call waiting tone for night service terminal	Continuous	Call waiting tone
Override tone	Continuous	Override tone

10.9 Portugal

10.9.1 Bothway Main Station Line without DID with Pulse Dialling or DTMF Signalling (TMANI).

Two-wire main station line to the central office for voice and signalling (HKZ).

The line can be operated on a

- bothway
- o/g only
- i/c only

basis.

Signalling method: pulse dialling or DTMF

I/c traffic always results in attendant intercept.

Features:

- Incoming traffic
  – Seizure (ringing ac)
  – Answering
  – Forward release
  – Backward release
- Outgoing traffic
  – Seizure
  – Dialling
  – Forward release
  – backward release
  – 12-kHz charge reception

Forward or backward release can take place on the central office side by "release by the central office through polarity reversal" or "release by the central office through feed interruption".

10.9.2 Analog tie traffic

Bothway Tie Trunk with AC Signalling for Pulse Dialling or DTMF Signalling (TMBC)

Two-wire bothway tie trunk for voice and signalling.

Signalling is by means of a 50-Hz ac voltage.

Signalling method: pulse dialling or DTMF

10.9.2.1 Bothway Tie Trunk with E&M (TMEW2)

Signalling takes place with DC voltage via the control wires E and M.

Dialling can alternatively be implemented using 50-Hz pulses or DTMF signalling.

10.9.3 Audible Tones for Portugal

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone	425	Continuous tone
Busy tone	425	**160**/480 bzw. **500**/500
ATB tone	425	**250**/250
Internal dial tone	350+440 superimposed	Continuous tone

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse/Pause [ms]	Level (dBm0)
Internal dial tone	1	350+440 superimposed	Continuous tone	-5
External dial tone	2	425	Continuous tone	-5
Ringback tone	4	425	**1000**/4000	-5
Busy tone	5	425	**160**/480	-5
Override tone	6	425	**200**/300/**200**/1300	-13.5
Call waiting tone	7	425	**100**/1000	-13.5
Special dial tone	3	425	**200**/300/**200**/300/**200**/800	-5
NU tone	8	950/1400/ 1800	Triple tone **332-332-332**/1000	-12
Data call tone	9	1300	**600**/1800	-1
Conference auxiliary tone	24	425	Continuous tone	-13.5

Tone	Tsl	Frequency [Hz]	Pulse/Pause [ms]	Level (dBm0)
LCR	26	1800	**340**/200/**340**/200/**340**/1000	- 5

Special tone sequences:

Tone	Sequence	derived from
Conference tone	200 ms 300 ms 200 ms Continuous	Conference auxiliary tone Pause Conference auxiliary tone Silence
Call waiting tone	Continuous	Call waiting tone
Call waiting tone for night service terminal	Continuous	Call waiting tone
Override tone	Continuous	Override tone

10.9.4 Discriminating Ringing

The system has three different call signals:

- Internal call signal, used for dialled or extended internal calls
- Trunk call signal, used for dialled or extended trunk calls
- Special call signal, used when an initiated callback is activated at the A-user end (the A-user has initiated the callback feature; at the end of the B-user's call, he is called back with the special call signal, to distinguish the call from an internal call).

Technical data

Frequency	22 ... 28 Hz	HiPath: 25 Hz)
Voltage	60 ... 90 V	HiPath: V_{rms}=75 V)

There are no rules governing the pulse/pause ratio of the call signals. The first ringing pulse should be transmitted up to one second after the connection has been established.

The following nominal values are implemented in HiPath 4000:

	Frequency (Hz)	**Pulse**/Pause (ms)	Voltage
Internal call signal	25	**1000**/4000	75V
Trunk call signal	25	**400**/200/**400**/4000	75V
Special call signal	25	**200**/200/**200**/200/**200**/4000	75V

10.10 Netherlands

10.10.1 Specific Features

10.10.1.1 Detection of the 1st and 2nd Trunk Dial Tone, and Forwarding to the User

For outgoing trunk calls the PABX recognizes the first and second trunk dial tone.

Forwarding to the user takes place as follows:

- first trunk dial tone to the user is injected as a simulated 425-Hz tone.
- second trunk dial tone (can be set up as an alternative per system)
  - correctly timed injection to the user
  - without injection to the user.
  - without second exchange dial tone evaluation.

As of HiPath 4000 the exchange tone detection procedures just described are modified to suit the different requirements of the Netherlands PTT which come into effect on April 1, 1994:

- The new exchange dialling tone will be a continuous 425-Hz tone
- The second exchange dial tone evaluation is no longer required.

10.10.1.2 Register Recall into the Public Network for Incoming Trunk Calls

With "register recall" into the public network an internal user is given the opportunity, during the call phase, to use specific public services offered by the exchange. The services are selected by suffixing a code.

When a service has been selected, signalling on the trunk proceeds as follows:

- the PABX sends the "register recall" signal as the line signal to the exchange,
- the exchange makes a register available for accepting the subsequent service code and acknowledges this by applying the 1st trunk dial tone (150 Hz),
- the PABX transmits the (max. 4-digit) service code in the form of a series of DTMF signals to the exchange.

The only service currently offered by the exchange is "Tracing an external user in incoming trunk traffic".

"Register recall" is only permissible in the case of trunks with DTMF signalling.

The register recall function

- can be set up on an system-specific basis in HiPath 4000,
- is only permissible when exchanges have been suitably equipped for this function.
- is only implemented for incoming exchange traffic

10.10.1.3 Call Tracing of an External User for Incoming Trunk Calls

This feature is only permissible for incoming trunk calls during the call phase and can be used by the following PABX users:

- Attendant console,
- Night answer extension,
- PABX extension user.

Call tracing orders are registered automatically in the PABX on the service terminal.

The following data are registered:

- Date and time of the start of call tracing,
- Call number of the internal user,
- Number of the trunk (hardware PEN),
- Additional text information (if applicable).

Call tracing classmark

It is an essential condition of using the call tracing feature that the PABX user has the call tracing classmark (can be set up with a MML command; no privileged code required for EMML).
Generally, the attendant console and night answer extension are classmarked for call tracing.

Exchange trunks can be configured with or without malicious call tracing
For each exchange trunk group configured with malicious call tracing, a specific number of trunks must remain free of malicious call trace jobs in order to avoid trunk group blocking (this is AMO-defined).
Once the limit value is reached, the system will reject all further malicious call trace requests for that trunk group.

There are the following two feature variants:

- Call tracing **with** installed register recall function,
- Call tracing **without** installed register recall function.

"Call tracing with installed register recall function" is only permissible in the case of trunks with DTMF signalling.

Restrictions

Call tracing is not possible in the HiPath 4000 network (the call tracing classmark cannot be transmitted throughout the network).

If the HiPath 4000 system carries out a soft/hard restart, lines (circuits) held for call tracing are released.

The max. number of trunks simultaneously held for call tracing can be set and changed by means of an AMO. Each time a call tracing request is made, the PABX checks whether this number has been reached. If it has, the internal user's order is negatively acknowledged.

The max. number of PABX users classmarked for call tracing can be set and changed by means of an AMO

Sequence of operations in the user call tracing procedure

The call tracing procedure can only be activated by the internal user during the call phase, in the following manner:

Call tracing from analog terminals

Initiate consultation by pressing the signal key (flash or ground key) and dial the call tracing code. The call tracing code can be set up on a system-specific basis for all PABX users (number of digits: 2 or 3). During the holding procedure the waiting status is signalled to the external user not by music nor an announcement.

Acknowledgments

If call tracing is accepted by the PABX, "positive acknowledgment tone" is applied.
Duration: generally 3.5 s, can be set by administration and maintenance.

If call tracing is not accepted by the PABX, "negative acknowledgment tone" is applied.
Duration: generally 3.5 s, can be set by administration and maintenance.
After the acknowledgment tone time the connection is automatically switched through to the exchange.

Call tracing procedure for digital terminals

Dial the call tracing code either during the call or by pressing the repertory/destination key set up for it; at the attendant console only by pressing the call tracing key. The connection to the outside party is not interrupted, the internal party remains in the call status.

Acknowledgments

for digital terminals (except for attendant console), on the display as follows:

- Text display "CALLER TRACED" in the local language, if call tracing is accepted by the PABX.
- Text display "NOT POSSIBLE" or "PLEASE REPEAT" in the local language, if call tracing is not accepted by the PABX.

for the attendant console on the screen as follows:

- Text display "STORED" in the local language, if call tracing was accepted by the PABX.
- Text display "NOT POSSIBLE" or "PLEASE REPEAT" in the local language, if call tracing is not accepted by the PABX.

A call tracing order accepted by the PABX is immediately registered on the service terminal.

10.10.1.4 Signalling on the Trunk

Depending on the call tracing variant established in the PABX (call tracing with/without register recall), the sequence of signalling events on the trunk is as follows in the case of:

- Call tracing with register recall
  - The PABX sends the "register recall" signal as the line signal (90 - 130 ms pulse) to the exchange.
  - The exchange acknowledges the readiness of a digit input register by applying the 1st trunk dial tone, which is evaluated by the PABX.
  - After evaluating the 1st trunk dial tone, as a request signal the PABX transmits the signal sequence for call tracing (*39#) to the exchange. The signal sequence meets the conditions for pushbutton dialling as per ITU Q.23 and CEPT recommendation T/CS 46-02.
- Call tracing without register recall
  - The PABX only sends the call tracing pulse as the line signal (100 - 200 ms pulse) with signalling code such as "register recall" to the exchange.
  - There is no acknowledgment of the call tracing pulse from the exchange.

"Traced" trunks are held in the exchange and released from there after the calling user has been identified.

If the PABX user goes on-hook after issuing a call tracing order, the connection is released internally from the PABX (by the circuit to the internal user). The trunk remains in the trace hold state until the forward release signal is received by the exchange. The trunk is then enabled by a clearing signal.

Printout of call tracing data

A call tracing order accepted by the PABX is immediately registered on the service terminal.

If, however, the exchange releases a held trunk connection within a specified period (e.g. 60 s) after the PABX user has gone on-hook, a non-executed call tracing order in the exchange is assumed. This is immediately registered in the PABX by a second printout at the service terminal with the following parameters:

- Date and time of the release of the held call by the exchange,
- Call number of the PABX user (as an allocation point for the 1st call tracing printout),
- Supplementary text "RELEASE DESPITE CALL TRACING" (in English).

10.10.1.5 Reanswering by PABX for Incoming Trunk Calls

This feature can only be used for incoming trunk calls and results in an automatic call (recall) at the attendant console if the PABX user is first to go on-hook and is not released by the exchange within a specified time limit (6 - 10s, settable) (with forward release signal). When the attendant enters, the "reanswering" line signal is sent to the exchange.

A recall at the attendant console is only possible once. A further recall is prevented by the PABX by releasing the connection to the exchange (with clearing signal).

When a call tracing order has been issued, a recall at the attendant console is suppressed.

10.10.2 Analog Exchange Traffic

Bothway Exchange Interface without DID (HKZ) (TMANI)

Signalling method:pulse dialling or DTMF

10.10.3 Bothway Trunk Calls for ALS 70 D Signalling (DIU-CAS)

- With DID for DTMF for incoming calls,

- without DID for DTMFfor incoming calls with attendant intercept.

The digital trunk circuit can be set up for

- one-way operation (only incoming/outgoing),
- bothway operation.

Features:

1. Incoming calls
   - Idle state
   - Seizure
   - Seizure acknowledgment
   - Ringing current clock
   - Ready
   - Dialling
   - End-of-dialling
   - Answer
   - Rering
   - Tracing
   - Clearing signal
   - Exchange call rering
   - Forward release
   - Barring
2. Outgoing calls
   - Idle state
   - Seizure
   - Seizure acknowledgment
   - Dialling
   - End-of-dialling
   - Call charge reception
   - Rering
   - Forward release
   - Backward release

- Barring

The ALS70 digital line signalling systems is implemented on the basis of the DIU-CAS. The signalling is channel-linked digital 2-bit user signalling for coupling a digital extension (PABX) to a digital exchange by means of PCM transmission at 2 Mbit/s in accordance with ITU recommendations G.703, G.704, G.705, G.732, G.736, Q.421, Q.424.

ALS70 digital signalling is used for one-way (incoming/outgoing) or bothway connections and employs the two signalling systems pulse dialling and DTMF (the latter in accordance with ITU recommendation Q.23 and CEPT recommendation T/CS 46-02) with the following combinations:

	outgoing	incoming
1	pulse dialling	pulse dialling
2	pulse dialling	DTMF
3	DTMF	pulse dialling
4	DTMF	DTMF

Incoming calls without DID

For incoming calls without DID there are two variants for activating call signalling at the attendant console of the PABX

- exchange only sends seizure signal,
- exchange additionally sends ringing current clock signal after the seizure signal.

Both variants are implemented as alternatives.

With the ringing current clock signal, the bf-bit periodically changes in the ringing cadence as follows:

Duration	Signal
1 s	bf = 1
4 s	bf = 0

Up to April 1, 1994, telephone exchanges in the Netherlands used two different exchange dial tones, which were appropriately catered for in the HiPath 4000 software. As of this date, the Netherlands PTT changed their requirements to only 1 exchange dial tone, so that the second exchange dial tone evaluation is no longer required.

- TThe new exchange dialling tone is a continuous 425-Hz tone
- The second exchange dial tone evaluation is no longer required.

Ready-to-dial condition for outgoing trunk calls

With an outgoing call, the ready-to-dial condition (readiness to accept dial information) is signalled by the exchange by the application of a dial tone. This dial tone is detected by the PABX.

10.10.4 Audible Tones for the Netherlands

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone	425	Continuous tone
Busy tone	425	**220...400**/200...400
Internal dial tone	425	**200**/300/**200**/300/**200**/800

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse/Pause [ms]	Level (dBm0)
External dial tone	2	425	Continuous tone	-3
Ringback tone	4	425	**1000**/4000	-3
Busy tone	5	425	**500**/500	-3
Override tone	6	425	**200**/300/**200**/1300	-14
Call waiting tone	7	425	**100**/1900	-14
Special dial tone	3	425+400	Continuous tone	-3
ATB tone	11	425	**250**/250	-3
Positive acknowledgement tone				
Internal dial tone	1	425	**200**/300/**200**/300/**200**/800	-3
NU tone	8	950/1400/ 1800	Triple tone **340-340-340**/1000	-3
Data call tone	9	1300	**600**/1800	1
Conference auxiliary tone	24	425	Continuous tone	-14
LCR	26	1800	**340**/200/**340**/200/**340**/1000	- 3

Special tone sequences:

Tone	Sequence	derived from
Conference tone	200 ms 300 ms 200 ms Continuous	Conference auxiliary tone Pause Conference auxiliary tone Silence
Call waiting tone	Continuous	Call waiting tone
Call waiting tone for night service terminal	Continuous	Call waiting tone
Override tone	Continuous	Override tone

10.11 South Africa

10.11.1 Call Processing Features with MFC-R2 Signalling

Override when a PABX user is Busy

If long-distance calls are operator-assisted, the attendant at the long-distance exchange has the option of causing the connection to be intercepted at the attendant console of the PABX if the PABX user is busy.

The attendant intercept takes place as soon as the PABX has received the override signal from the exchange.

The requirements relating to this feature are identical to those for Finland MFC-R2

A-user Number Identification

The number of the A-user is identified during the call setup phase of outgoing trunk calls.

The exchange only requests the A-user number – by **once** applying the backward signal "A5" – after complete interrogation of the B-user number. The PABX responds to the request by transferring the A-user number; each transferred digit of the A-user number is acknowledged by the exchange with "A5".

The A-user number consists of an up to 5-digit long extension-user number.

10.11.2 Analog Trunk Calls

Incoming, One-way Trunk Calls with DID (BPO) (TMLSF/TMLRB)

Two-wire main station line for voice and signalling (loop in signalling) (TMLSF), feeding from the PABX.

Signalling method:pulse dialling or DTMF

Features:

- idle state
- Seizure
- Seizure acknowledgment
- Start of pulsing
- Dial tone
- Dialling

- End-of-dialling
- Answering
- Forward release
- Backward release after answering
- Barring by the PABX

Bothway Trunk Calls without Direct Inward Dialling (MSI) (TMANI)

Two-wire main station line for voice and signalling (ring in world/loop out) (TMANI), Outgoing seizure with end-of-dialling simulation.

Signalling method:pulse dialling or DTMF

Features:

- Idle state
- Seizure
- Seizure acknowledgment
- Dialling
- End-of-dialling
- Answering
- Forward release
- Backward release
- Barring by the exchange
- Call charges
- Reseizure lockout

10.11.3 Bothway Trunk Calls with DID (MFC-R2 as per Q.421) (DIU-CAS)

The bits in signalling channel 16 of the PCM30 connection are used for line signalling (as per ITU G.732/Q.421). Only bits a and b are used in both directions; bits c and d are fixed to c=0 and d=1

The line signals are sent either as a continuous signal or as a pulse and, with the exception of charge pulse, release request, and override, are identical in both directions.

Module:DIU-CAS

Connect to EWSD as public exchange.

Signalling method:MFC-R2

Features:

- Idle state
- Seizure
- Seizure acknowledgment
- Answering
- Call charge pulse (outgoing traffic)
- Clearing signal
- Release request (outgoing traffic)
- Release
- Busy override (if PABX user is busy for incoming calls)
- Backward blocking
- Duration of call charge pulse
  Pulse: 200ms, Tolerance 100 ms to 1s
  Pause:75ms (min.)
- busy override
  - as a continuous signal on bit b; attendant intercept does not take place until the signal has been applied for ≥700 ms and has thus been reliably detected,
  - as a pulse on bit a; start and end of override is received as a 150-ms pulse on the a-bit.

10.11.4 Audible Tones for South Africa

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone	367+433	Continuous tone
Busy tone	400	**500**/500
ATB tone	400	**250**/250
Internal dial tone	400	Continuous tone

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse/Pause [ms]	Level (dBm0)
Internal dial tone	1	400	Continuous tone	-7

Tone	Tsl	Frequency [Hz]	**Pulse**/Pause [ms]	Level (dBm0)
External dial tone	2	367+433	Continuous tone	-7
Ringback tone	4	367+433	**400**/200/**400**/2000	-7
Busy tone	5	400	**500**/500	-7
Override tone	6	400	**200**/300/**200**/1300	-9
Call waiting tone	7	400	**100**/15000	-11
Special dial tone	3	400	**40**/40	-7
NU tone	8	400	**2500**/500	-7
Data call tone				
ATB tone	11	400	**250**/250	-7
Conference auxiliary tone 2	23	400	Continuous tone	-9
Conference auxiliary tone	24	400	Continuous tone	-11
LCR	26	1800	**340**/200/**340**/200/**340**/1000	- 7

Special tone sequences:

Tone	Sequence	derived from
Conference tone	200 ms 300 ms 200 ms Continuous	Conference auxiliary tone Pause Conference auxiliary tone Silence
Call waiting tone	Continuous	Call waiting tone
Call waiting tone for night service terminal	Continuous	Call waiting tone
Override tone	Continuous	Override tone

10.12 China

10.12.1 Call Processing Features

10.12.1.1 Release Control With Reanswering And Rering

In the Chinese telephone network connections can be released by

- the calling or called party (first party control) or
- the calling party (calling party control) or
- the called party (called party control).

This applies to Anate, Digite and Attendant Console users.

The relevant party is a PABX user

If a PABX user is speaking to an external partner in the public network who has control over this connection and is the first to go on-hook, he has the possibility of re-establishing the connection by going off-hook within a specific period or prior to his partner's release.

These partners can be:

- external incoming
  - callers
  - public exchange operator
- external outgoing:
  - user with a malicious call tracing circuit
  - public reservation and information positions
  - special services (attendant, fault report office, police, fire service, etc.)

The call types can be recognised in the case of:

- incoming seizures by the Group II signal, and in the case of
- outgoing seizures by external dial analysis.

The special status 'restricted' has been introduced for voice terminals which take one of the calls mentioned. In this condition voice terminals can only go on or off hook (reentering the connection). The connection is maintained until released at the external side. No features are permitted in this condition except for call tracing and long-distance attendant override on local calls.

The relevant party is an external party

Forward release (calling party control) is only effective on outgoing calls. Backward release from the central office is acknowledged with delayed forward release to allow the waiting external parts to reenter the connection or to bridge brief interruptions due to manual call servicing. The Release Control feature must be activated via administration and maintenance order, and cannot be activated manually.

1. The Release Control feature was realised in HiPath 4000 for the CSN1 protocol. As of HiPath 4000, this feature also operates in CorNet-NQ networks.
2. The control over the connection is held by the party with the release control authorisation.
   - If a party with the release control feature hangs up (goes on-hook), the following takes place:

- If the partner does not have release control authorisation, the connection is released immediately.
- If the partner does have release control authorisation, the change in the device status is indicated.

- If a party without the release control feature hangs up (goes on-hook), the following takes place:
  - If the partner does not have release control authorisation, the connection is released immediately.
  - If the partner does have release control authorisation, the change in the device status is indicated.

The possibility of reanswering **depends on the partner's release control authorisation**:

- If a party with the release control feature hangs up (goes on-hook), the following reanswer options are available:
  - If the partner does not have release control authorisation, the party with the release control authorisation can reanswer by lifting the receiver again.
  - If the partner also has release control authorisation, either party can reanswer, provided the other is off-hook.
- If a party without the release control feature hangs up (goes on-hook), the following reanswer options are available:
  - If the partner does not have release control authorisation, reanswering is not possible.
  - If the partner does have release control authorisation, reanswering is possible.

In all cases, parties without release control authorisation and rering authorisation are monitored for a specific time after the partner has hung up. The timer is stopped as soon as the user lifts the handset again. Once the timer has run down, the connection is finally released.

3. When a connection partner goes on-hook, the system immediately checks his release control authorisation.
   - Users with release control authorisation will not receive any special signalling (e.g. tones, announcements, display outputs) if the connection partner goes on-hook.
   - If the connection partner has release control authorisation, users without release control authorisation are put into ringing state when they go on-hook, without a ringing signal being applied. This means that the line

remains ’open’, i.e. the user is returned to the connection partner if he lifts the handset (goes off-hook) during the monitoring period. The telephone display will show ’PARTNER WAITING’ during this time.

10.12.1.2 Automatic Override, Toll Call Offering TCO

If a long-distance attendant or a remote HiPath user dials a user in DID who is busy with an internal or local call, this call is automatically overridden with a tone. After the attendant has informed the required party about a waiting long-distance call, that party goes on-hook if he wishes to accept it. When the relevant party goes off-hook again he is connected to the long-distance attendant or the HiPath user, but without override tone. If the party does not go off-hook again, the long-distance attendant can rering.

- Trunk calls have total override protection.
- Automatic override is only carried out on connections set up in CSN1 networks, or connections which branch into CSN1 networks. The feature is rejected, if the destination PABX is in a CorNet-NQ network.
- If the dialled user is engaged in a consultation call, automatic override is not possible. However, HiPath 4000 users can initiate automatic override **for** consultation calls.

10.12.1.3 Rering or Ringback from the Attendant

If a PABX user goes on-hook from a connection with the attendant and the attendant does not yet wish to terminate this connection, depending on the direction of the previous call the attendant can implement either rering or ringback to the PABX user.

10.12.1.4 Override Condition of the Attendant Console

The attendant is only able to override internal or local calls. Trunk calls have total override protection.

10.12.1.5 Manual Override

In addition to automatic override, manual override for networks has also been realised for China as of HiPath 4000.

- For incoming connections, this feature can only be initiated by attendants, by pressing the appropriate key. The feature functions for incoming connections from CSN1 networks and CorNet-NQ networks (from HiPath 4000).

- For outgoing connections, the feature can only be used within CorNet-NQ networks. If a breakout occurs into a CSN1 network, the feature is rejected.
- If a transit node in a network route has no free outgoing trunks, the caller will receive the ATB tone. If the transit node is unable to transfer the ’manual override’ signal, the caller will receive the busy tone.
- Manual override is possible on busy Digites, Anates, or attendant telephone numbers.
- Manual override is only possible if the dialled line has no override protection, and is part of a basic two-party connection.
- Manual override is rejected in the following situations:
  - if the called line is not in talking state (on hold, dialling, ringing etc.),
  - if devices with special data protection facilities are connected:
    data terminals,
    fax and vtx devices,
    paging/code-calling sytems,
    public address systems,
    dictation equipment,
    voice announcement units,
    VMS,
    entrance telephone,
  - if either of the parties engaged in the connection has override protection (either A-user or B-user),
  - if a connection is already being overridden,
  - if one of the parties already has a call waiting,
  - if the called line is also attempting to initiate override or call waiting,
  - if the called line already has a held call,
  - if the called line is engaged in a consultation call,
  - if the called line is participating in a conference call,
  - if the called line is connected to the VMS.
- If manual override is not possible, the caller will hear a busy tone.
- If manual override is enabled, all parties concerned will hear the override tone when override is initiated.
- The following actions lead to a talking connection of the overriding caller with the called party:
  - Called party hangs up. This causes a rering, and the overriding party will hear the ringback tone.

– The called party puts his original call on hold.
  the called party can toggle between the overriding caller and the held caller
  if the called party hangs up while the original caller is still on hold, a rering is initiated

– Called party releases original call by pressing a key:
  Disconnect/Delete key. The original call is disconnected, and the called party is connected with the overriding caller.
  Called party presses disconnect/delete key during a consultation call while an additional party is attempting manual override: called party is connected with override caller.
  An overriding call has priority over a second held call, if the called party presses the disconnect/delete key.

10.12.1.6 R2-MFP Signalling

Apart from the MFC-R2 signalling method, the R2-MFP signalling method is also provided for China as of HiPath 4000.

The following two MFP signals are applied:

- MFP1
  - for transit routes between two nodes,
  - for incoming and outgoing traffic (not bothway),
  - as pulse-signals for forward and backward signalling,
  - for the release control feature (calling party only).
- MFP2
  - for simple tie-traffic, also on transit routes,
  - for incoming traffic only,
  - as pulse-signals for forward signalling,
  - for the release control feature (calling party only).

For the MFP-signals, the same frequency combinations and tolerances apply as for MFC-R2.

The following pulse/pause ratios are defined:

- Pulse

Forward:	100 ± 20ms Range 50 - 500 ms

Backward:	max. 10-60 s min. 40ms

- Pause

Forward:	100 ± 20ms Range 50 - 500 ms
Backward:	same as max. pause length

When the maximum pulse or pause receive time is exceeded, an existing connection will be cleared down.

10.12.1.7 Line Signalling for Digital and Analog Exchange Lines

For HiPath 4000 exchanges in the public network, and for the Railway Network in China (CSN1 networking protocol), the following interfaces are available:

- Digital signalling: PCM30-interface (DIUN2)
- Analog signalling: 4-wire voice-frequency interface (2600-Hz-inband-signalling via TMSFP).

Apart from the line signals for connection setup and release, the signalling protocols allow the following feature signals, depending on the type of signalling:

- Forced Break
  Operator feature. If an operator wishes to set up an outgoing connection to a busy subscriber outside his own PABX, the operator's PABX can enforce a connection between the operator and the subscriber by means of this signal.
- Rering
  Operator feature. If a called subscriber outside an operator's PABX hangs up, the operator's PABX will initiate a rering via the subscriber's PABX by means of this signal.
- Ringback
  Operator feature. If a subscriber outside an operator's PABX calls the operator and then hangs up, the operator's PABX can initiate a ringback (callback) via the subscriber's PABX by means of this signal.
- Return Call to Operator
  Subscriber feature. If a subscriber is called by an operator of a remote system, who then hangs up, the subscriber can initiate a return call to the operator by means of this signal.
- Malicious Call Tracing
  Subscriber feature. A called subscriber with the appropriate authorisation can initiate a call trace in the remote PABX by means of this signal. This only works in digital signalling systems.

- Metering
  Call charge pulses are transferred back to the PABX from which the call originated by means of this signal. This only works via CSN1 networks with digital signalling systems.

10.12.2 Analog Exchange Traffic

10.12.2.1 Bothway Trunk Traffic with DID as per MFC-R2 (CSN1) via DIU-CAS with KZU

Signalling method:pulse dialling

Signalling conversion for CSN1 takes place via SC MUX (S42023-A757-A209) from ÖN ÜB in the local exchange. Connection to the DIU-CAS is via PCM30.

10.12.2.2 Bothway Main Station Line with 16-kHz Charge Pulse Reception and 25-Hz Ring Detection (TMANI)

2-wire line with silent reversal signalling. Silent reversal permits detection of voltage reversal also in the idle condition.

Signalling method:pulse dialling or DTMF

Features (as per Q.421):

- Idle
- Seize
- Seizure acknowledgment
- Answer
- Forward release
- Backward release
- Reseizure lockout
- Rering

Special criteria:

- Guard time for reseizure lockout can be set from 3 to 60 s
- Release in the event of no answer in outgoing traffic after 90 s
- Delayed enablement in the event of backward release after answering in outgoing traffic 90s

10.12.3 Analog Tie Traffic

10.12.3.1 Bothway Tie Traffic with E&M Signalling via DIU-CAS with KZU

Signalling conversion for CSN1 takes place via SC MUX (S42023-A757-A209) from ÖN ÜB in the local exchange. Connection to the DIU-CAS is via PCM30.

10.12.3.2 Incoming Tie Traffic with MFC-R2 (CSN1)

Signalling conversion for CSN1 takes place via SC MUX (S42023-A757-A209) from ÖN ÜB in the local exchange. Connection to the DIU-CAS is via PCM30.

10.12.3.3 Outgoing Tie Traffic with MFC-R2 (CSN1)

Signalling conversion for CSN1 takes place via SC MUX (S42023-A757-A209) from ÖN ÜB in the local exchange. Connection to the DIU-CAS is via PCM30.

10.12.4 Digital Trunk Traffic

Incoming Trunk Traffic with DID as per MFC-R2 (CSN1) via DIU-CAS

Signalling conversion for CSN1 takes place via SC MUX (S42023-A757-A209) from ÖN ÜB in the local exchange. Connection to the DIU-CAS is via PCM30.

Features defined in addition to those in Q.421 are:

- Rering
- Release check (reanswering)
- Reseizure lockout

Outgoing Trunk Traffic as per MFC-R2 (CSN1) via DIU-CAS

Signalling conversion for CSN1 takes place via SC MUX (S42023-A757-A209) from ÖN ÜB in the local exchange. Connection to the DIU-CAS is via PCM30.

Features defined in addition to those in Q.421 are:

- Rering
- Charges (recording on charge pulse or time basis)
- Call tracing
- Release check (reanswering)

- Reseizure lockout

10.12.5 Digital Tie Traffic

Incoming Tie Traffic as per MFC-R2 (CSN1) via DIU-CAS in the Local Exchange Area

Signalling conversion for CSN1 takes place via SC MUX (S42023-A757-A209) from ÖN ÜB in the local exchange. Connection to the DIU-CAS is via PCM30.

Features defined in addition to those in Q.421 are:

- Rering
- Release check (reanswering)
- Reseizure lockout

Outgoing Tie Traffic as per MFC-R2 (CSN1) via DIU-CAS in the Local Exchange Area

Signalling conversion for CSN1 takes place via SC MUX (S42023-A757-A209) from ÖN ÜB in the local exchange. Connection to the DIU-CAS is via PCM30.

Features defined in addition to those in Q.421 are:

- Rering
- Charges (recording on charge pulse or time basis)
- Call tracing
- Release check (reanswering)
- Reseizure lockout

10.12.6 Signalling Method

The pulse dialling and DTMF signalling methods are used as well as the MFC-R2 method as per ITU Q.440-442.

The signals employed differ partly from those specified in Q.440-442:

10.12.7 Digital Data and Voice Transmission

Network-wide transmission via PCM30 highways with 64kBit/s. Since the German COCOM regulations (export license regulations) do not allow data transmission in China to be implemented via DIUT2 (ISDN) interface units, DIUN2 interface units will be used.

The B-channels of the DIUT2 are configured to allow voice transmission only.

10.12.8 Hörtöne China

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse/Pause [ms]
Exchange dial tone	400	Continuous tone
Busy tone	450	**350**/350
ATB tone	450	**700**/700
Internal dial tone	400	Continuous tone

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse/Pause [ms]	Level (dBm0)
Internal dial tone	1	400	Continuous tone	-9
External dial tone				
Ring back tone	4	450	**1000**/4000	-9
Busy tone	5	450	**350**/350	-9
Override tone	6	450	**200**/200/**200**/600	-19
Call waiting tone	7	450	**400**/4000	-19
Special dial tone	3	400	**400**/40	-9
NU tone	8	450	**100**/100/**100**/100/**100**/100/ **400**/400	-9
ATB tone	11	450	**700**/700	-9
1st ring back tone	12	450	Dauerton	- 9
Conference tone	15	1400	**400**/10000	- 19
Auxiliary conference tone 2	23	450	Continuous tone	- 19
Auxiliary conference tone	24	1400	Continuous tone	- 19
LCR	26	1800	**340**/200/**340**/200/**340**/1000	-9

Special tone sequences:

Tone	Sequence	derviced from
Conference tone	Continuous	Conference tone
Call waiting tone	Continuous	Call waiting tone
Call waiting tone for night service terminal	Continuous	Call waiting tone
Override tone	Continuous	Override tone

10.13 Argentina

10.13.1 Scheduled Injection of Trunk Dial Tone

The dial tone is supplied to the station by means of timed injection. This procedure is necessary, since the ready-to-dial condition of public exchanges in Argentina does not occur - under certain circumstances - until minutes after the request.

10.13.2 Two-Way Trunk Traffic (Main Station Interface) with Call Charge Registration 50Hz (TMANI)

Two-wire main station line for voice and signalling.

Dialling method:Pulse dialling or DTMF

10.13.3 Two-Way Trunk Traffic with Loop Signalling (TMANI)

Two-wire, two-way DID tie trunk to the remote system for voice and signalling.

10.14 Malaysia (ASEAN)

10.14.1 Analog Trunk Traffic

10.14.1.1 Two-Way Trunk Traffic (TMANI)

Without DID, with loop start / silent reversal signalling.

Dialling method:Pulse dialling, DTMF

10.14.1.2 Two-Way Trunk Traffic (TMANI)

Without DID, with main station interface, call charge registration 16 kHz.

Dialling method:Pulse dialling, DTMF

10.14.1.3 Incoming Trunk Traffic (TMLSF)

With DID, loop-in signalling, feeding from the PABX.

Dialling method:Pulse dialling, DTMF

10.14.2 Audible Tones for Malaysia

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone	400+450	Continuous tone
Busy tone	425	**500**/500
ATB tone	425	**500**/250
Internal dial tone	425	Continuous tone

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse / Pause [ms]	Level [dBm0]
Free channel		0		
Internal dial tone	1	425	Continuous tone	- 5
External dial tone	2	425	Continuous tone	- 7
Special dial tone	3	425+400	Continuous	- 5
Ringback tone	4	425	**400**/200/**400**/2000	-10
Busy tone	5	425	**500**/500	- 10
Override tone	6	425	**200**/200/**200**/5000	-16
Call waiting tone	7			
NU tone	8	425	**2500**/500	-10
Pos. acknowledgement tone	10	425	**40**/40 for 1000 ms, then silence	-5
ATB tone	11	425	**500**/250	-10
Conference tone	12	425	Continuous	-10
Neg. acknowledgement tone	14	425	**100**/200	-10
Conference auxiliary tone 2	23			
Conference auxiliary tone	24	425	Continuous	-16
LCR	26	1800	**340**/200/**340**/200/**340**/1000	- 10

Special tone sequences:

Tone	Sequence	derived from
Conference tone	1000 ms Continuous	Conference tone Silence
Call waiting tone	100 10,000 100 Continuous	Conference tone Free channel Conference tone Silence
Call waiting tone for night service terminal	100 100 100 10,000 100 100 100 Continuous	Conference tone Free channel Conference tone Free channel Conference tone Free channel Conference tone Silence
Override tone	200 200 200 Continuous	Conference auxiliary tone Free channel Conference auxiliary tone Override tone

10.15 Singapore (ASEAN)

10.15.1 Analog Trunk Traffic

10.15.1.1 Two-Way Trunk Traffic (TMANI)

Without DID, with loop start / silent reversal signalling.

Dialling method: Pulse dialling, DTMF

10.15.1.2 Two-Way Trunk Traffic (TMANI)

Without DID, with main station interface, call charge registration 16 kHz.

Dialling method: Pulse dialling, DTMF

10.15.1.3 Incoming Trunk Traffic (TMLSF)

With DID, loop-in signalling, feeding from the PABX.

Dialling method: Pulse dialling, DTMF

10.15.1.4 Two-Way Trunk Traffic (TMANI)

Without DID, with silent reversal signalling, call charge registration 16 kHz.

Dialling method: Pulse dialling, DTMF

10.15.2 Audible Tones for Singapore

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone	400+450	Continuous tone
Busy tone	425	**750**/750
ATB tone	425	**250**/250
Internal dial tone	425	Continuous tone

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse / Pause [ms]	Level [dBm0]
Internal dial tone	1	400+450	Continuous tone	- 7
External dial tone	2	425	Continuous tone	- 5
Special dial tone	3	480+620	**250**/250	- 5
Busy tone	5	425	**750**/750	- 10
Override tone (call waiting tone)	6	425	**200**/200/**200**/5000	-16
Ringback tone	4	400+450	**400**/200/**400**/2000	-12
NU tone	8	425	**2500**/500	-10
ATB (conference entry)	11	425	**250**/250	-10
Conference tone				
Initial ringback tone	15	400+450	Continuous tone	-12
Conference auxiliary tone 2	23	425	Continuous tone	-10
Conference auxiliary tone	24	425	Continuous tone	-16
LCR	26	1800	**340**/200/**340**/200/**340**/1000	- 12

Special tone sequences

Tone	Sequence	derived from
Conference tone	4000 ms	ATB tone
Call waiting tone	Continuous	Call waiting tone
Call waiting tone for night service terminal	Continuous	Call waiting tone

Tone	Sequence	derived from
Override tone	200 ms 200 ms 200 ms Continuous	Conference auxiliary tone Pause Conference auxiliary tone Override tone

10.16 Thailand (ASEAN)

10.16.1 Specific Features

10.16.1.1 Direct Inward Dialling with/without Announcement

The HiPath 4000 administration and maintenance system allows two alternatives: on reaching the HiPath 4000 system, callers are immediately switched to the HiPath station number dialled, or receive an announcement asking them to dial the desired station number. In the latter case, the station number can also be entered immediately, i.e. without having to wait for the announcement to finish.

10.16.1.2 Announcement for Delayed Attendant Answering

Calls to the attendant console, which are not immediately answered, are switched to an announcement (’Announcement 1’) after a pre-defined time (6 - 12 seconds). Callers will hear music on hold for max. 30 seconds, after which ’Announcement 2’ is started, which is, in turn, followed by announcement 1 again. The number of ’announcement 1’ and ’announcement 2’ cycles can be defined. The attendant can answer the held call at any time and interrupt the cycle. If the attendant does not answer within the defined number of cycles, the call is released.

10.16.1.3 Special Dial Tone for Consultation Calls

When initiating a consultation call, users hear the conference tone instead of the standard dial tone.

This allows users to differentiate between

- Connection release due to adequate interrupt duration with subsequent re-seizure (positively acknowledged with internal/external dial tone) and
- Release attempt due to inadequate interrupt duration, e.g. with hook-flash causing accidental consultation call setup (negatively acknowledged with conference tone).

10.16.2 Analog Trunk Traffic

Two-Way Trunk Traffic (TMANI)

Without DID, with loop start / silent reversal signalling.

Dialling method:Pulse dialling, DTMF

Two-Way Trunk Traffic (TMANI)

Without DID, with main station interface, call charge registration 16 kHz.

Dialling method:Pulse dialling, DTMF

10.16.3 Digital Trunk Traffic

Bothway exchange traffic with 2-bit signalling via signalling timeslot 16 and MFC-R2 user-channel signalling (DIU-CAS).

10.16.4 Audible Tones for Thailand

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone	367+433	Continuous tone
Busy tone	400	**500**/500
ATB tone	400	**250**/250
Internal dial tone	425	Continuous tone

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse / Pause [ms]	Level [dBm0]
Internal dial tone	1	350+400	Continuous tone	-10
External dial tone	2	400+450	Continuous tone	-10
Special dial tone	3	350	Continuous tone	-10
Ringback tone	4	400	**1000**/3000	- 8
Busy tone	5	408+620	**500**/500	- 8
Override tone	6	440	**200**/200/**200**/4000	-16
Call waiting tone		440	**200**/30.000	-16
NU tone also negative acknowledgement tone and re-order Tone	8	400	**100**/100/**100**/100/**100**/100/ **300**/100	-10

Tone	Tsl	Frequency [Hz]	**Pulse** / Pause [ms]	Level [dBm0]
Positive acknowledgement tone also 'Service Set Tone'	10	440	Continuous tone	-10
ATB	11			
Conference tone also Special dial tone after hook-flash	12	350+440	**120**/120	- 10
Initial ringback tone	15	400	Continuous tone	- 8
Conference auxiliary tone	24	440	Continuous tone	-16
LCR	26	1800	**340**/200/**340**/200/**340**/1000	- 8

Special tone sequences:

Tone	Sequence	derived from
Conference tone	1000ms Continuous	Conference tone Silence
Call waiting tone	200 ms Continuous	Conference auxiliary tone Call waiting tone
Call waiting tone for night service terminal	200 ms Continuous	Conference auxiliary tone Call waiting tone
Override tone	200 ms Continuous	Conference auxiliary tone Override tone

10.17 Spain

10.17.1 Call Processing Features

10.17.1.1 Converting the Route Code Digits of the Public Network Numbering Plan to "Call Type" Codes for Outgoing Exchange Calls

The Spanish public telephone network has a fixed, 8-digit numbering plan:

X Y A B M C D U

X Y code for areas with 6-digit numbers (standard case)

X code for areas with 7-digit numbers (large cities, e.g. Madrid = 1

X, Y and the first digit of the subscriber number (Y or A) may have a value between 1 and 8.

For calls within the same area, only the 6 or 7 digits of the subscriber number are dialled. Calls to other areas are set up by dialling the route discrimination code "9" followed by the area code.

If the possible "Y" digit combinations are not all assigned, in areas with 7-digit subscriber numbers, these "Y" numbers can be used as an additional route discrimination code for sub-areas with 6-digit numbering plans, e.g. Avila (= 18), a suburb of Madrid (= 1).

In addition to the subscriber numbers, there are also 2-digit codes for bearer services, which begin with 0 to 6 or 8 to 9. Bearer services within the local area are reached by dialling the route discrimination code "0", followed by the service code (e.g. "003"). Bearer services in other local areas can only be accessed if they are allocated a code beginning with "0" or "9", i.e. the subscriber must dial theroute discrimination code "9", followed by the area code, followed by the service code (e.g. "9103" or "94703" accesses the same service in different areas).

International calls are set up by dialling the route discrimination code "07", followed by the international dialling code after the second 600-Hz-dialling tone (e.g. 07 44 for UK).

The register coding for DTMF signalling is as follows:

- The route discrimination codes "9" (toll calls) and "07" (international calls) are only converted to group II signals and not transmitted to the exchange as group I dialling information signals.
- The route discrimination codes "0" (local bearer services) and "1...8" (sub-area code) are both converted to group II signals and transmitted to the exchange as group I dialling information signals.

10.17.1.2 Answering Criterion Monitoring on Outgoing Call Setup

If no answering criterion is received from the exchange within 60 seconds after the end-of-dialling signal, the connection is released, and the caller hears the busy tone.

10.17.1.3 Release Control and Re-Answering

In exchange traffic, the calling party normally initiates a forward release at the end of the call. For backward release, a backward release criterion is sent to the calling party's local exchange and monitored here. Unless the called party answers again within 60 seconds, the connection is cleared down. If the calling party goes on-hook immediately after the called party, the connection is cleared down immediately.

Calls to bearer services are the exception to the rule. If a caller goes on-hook after receiving the answering criterion of a bearer service, an open-line dialling information criterion is sent down the line instead of the forward release criterion (Supervision a operadora). If the bearer service procedures have been completed (e.g. announcement is finished), the connection is cleared down immediately, regardless of whether or not the subscriber goes on-hook.

10.17.1.4 Digit Analysis of Out-Dial Information to the Public Network Exchange

Digit Analysis of Out-Dial Information to the Public Network Exchange. In order to discriminate between bearer service calls and to detect the route discrimination codes for the various call types, the dialled digits are analysed before being transmitted to the exchange. The route discrimination codes can be entered in the dial plan of the system via AMO.

The dialled number conversion (conversion of route discrimination codes) and the code block conversion for DTMF 2/5 signalling is included in the dial plans and alternate routing plans of the Least Cost Routing (LCR) system.

Calls to certain bearer services (e.g. emergency calls) are automatically switched via the trunk group configured for these routes by the LCR system, regardless of the callers' ITR group restrictions.

Since the LCR system is only implemented as an alternative to the existing external toll/code restriction system (EWAKO tables), both the "2nd exchange dialling tone" and "class-of-externals (COE) analysis" functions are taken into account in these tables as well.

If the international route discrimination code "07" is dialled by a PABX user, he or she will receive the 600-Hz second dialling tone from the PABX.

10.17.1.5 Digit Analysis of In-Dial Information for DID (DDI) Calls

For incoming exchange calls, the last 5 digits of a call number (e.g. BMCDU) are transmitted.

10.17.1.6 Individual Call Data Recording in the Public Network Exchange

This feature is fixed in the exchange and in the PABX, for each individual station user.

The call charge assignment option can be encoded in the group II signals for 2/6 register signalling (2 out of 6 frequencies with 15 signals):

- call charge assignment per trunk group (tarificación por bloque) as for 2/5 register signalling (2 out of 5 frequencies with 10 signals)
- call charge assignment per station (tarificación por línia).

For call charge assignment per station, the station line is identified by the exchange.

10.17.1.7 Call Charge Pulses from the Public Network Exchange

Irrespective of the call charge assignment option selected for individual call data recording in the exchange, call charge pulses can also be transmitted per call by the exchange (if implemented) for evaluation in HiPath 4000.

10.17.2 Bothway main station line without DID with pulse dialling or DTMF signalling (TMANI).

Two-wire main station line to the central office for voice and signalling (HKZ).

The line can be operated on a

- bothway
- o/g only
- i/c only

basis.

Signalling method: pulse dialling or DTMF

I/c traffic always results in attendant intercept.

Features:

- Incoming traffic
  - Seizure (ringing ac 25 Hz)
  - Answering

  - Forward release
  - Backward release
- Outgoing traffic
  - Seizure
  - Dialling
  - Forward release
  - backward release
  - 12-kHz charge reception

Forward or backward release can take place on the central office side by "release by the central office through polarity reversal" or "release by the central office through feed interruption".

10.17.3 Bothway tie trunk with E&M (TMEW2)

Signalling takes place with DC voltage via the control wires E and M.

Dialling can alternatively be implemented using 50-Hz pulses or DTMF signalling.

10.17.4 Outgoing and Incoming Exchange Connections with Transit Signalling

Digital exchange traffic is carried out via PCM 30 highways with channel-associated signalling and register dialling via the voice channels. Separate trunk groups are configured for incoming and outgoing traffic.

Digital exchange traffic is based on a transit signalling system via PCM30 CAS. The system consists of single-bit line signalling via continuous code (E&M) and the multi-frequency register signalling system SMF (Spanish Multi-Frequency, similar to the French SOCOTEL system). A further bit is used for transmitting call charge pulses to the PABX.

Normally, connection release is controlled by the calling party, i.e. as long as the calling party does not go on-hook, the called party can re-answer a call within one minute of a backward release (re-plugging time).

However, if one of the bearer services ("Servicios Especiales", such as police, ambulance, public network operator etc.) is called, the service line circuit controls the release without a time limit. The status of the caller (on-hook/off-hook) is indicated on the bearer service telephone. The "special service" telephone user cannot re-ring the caller. Connections to bearer services are detected via digit analysis in the PABX.

Depending on the type of public network local exchange, one of two register-signalling variants are used:

"2/5"	2 out of 5 frequencies, 10 signals:		
	700, 900, 1100, 1300, 1500 Hz	and base frequency	1700 Hz,
"2/6"	2 out of 6 frequencies, 15 signals:		
	700, 900, 1100, 1300, 1500, 1700 Hz	and base frequency	1900 Hz.

Both variants support the feature "conversion of route code digits to 'call type' codes".

The 2/6 register signalling variant also supports call charge assignment per station line (must be implemented in the exchange). The station line number is identified in the exchange for the purpose of call charge assignment only; no other identification is envisaged. Call charge pulses are still transmitted to the HiPath 4000 by the exchange for internal evaluation.

The register signalling options can be expanded via AMO. This allows for compatibility with the IBERCOM features planned for the future.

If the users are divided into several different groups (virtual PABXs/nodes), each group is assigned a specific trunk group for outgoing calls. A common trunk group is used for incoming exchange calls.

- Line signalling (outgoing and incoming)
  - Idle state
  - seizure
  - seizure acknowledgement
  - answering criterion
  - call charge pulse (incoming direction only)
  - signal to "special services" (outgoing direction only)
  - backward release
  - forward release acknowledgement
  - blocking by calling line
- Register signalling
  The Spanish Multi-Frequency register-signalling system (SMF)is based on a forced frequency passage system with dialled signal detection via the voice transmission channels (similar to the French SOCOTEL system). The code frequencies used correspond to those defined in the ITU Recommendation no 5 and SOCOTEL specifications: 700, 900, 1100, 1300, 1500, 1700, and 1900, Hz. The same register signalling variants are used for forward and

backward dignalling (2 out of 'n'). Each received signal is acknowledged with a base control frequency. All signalling sequences and the entire signal exchange are time-supervised.

Notes:

- Release monitoring
  - Exchange traffic release is carried out according to the "calling party release control" principle, with a time supervision of 60 seconds, i.e. when the called party goes on-hook, this is signalled down the line but does not lead to a connection release until the 60 seconds have elapsed. If the called party goes off-hook again during this period, both parties can resume their conversation where they left off. If the calling party goes on-hook, the connection is released immediately.
  - For outgoing calls to "special services", connection release is carried out according to the "called party release control" principle. The connection can be cleared down by forward release before the "special services" line answers. Once the line has answered, the connection can no longer be cleared down via forward release. If the calling party goes on-hook first, a cyclic pulse is transmitted down the line from the calling party's end in order to indicate this to the special services line. The pulse signal disappears when the calling party comes back on the line (i.e. goes off-hook again), and the connection is once more in talking state. The connection is terminated by the special services line on completion of the call.
  - The cyclic pulse signal must fulfil the following time requirements:
    1-pulse:100 ±20 ms
    0-pulse: 400 ±20 ms
- Call charge pulses from the exchange
  - Call charge pulses can only be received for outgoing exchange connections.
  - Call charge pulse detection begins when the called party answers.
  - Call charge pulses must have the following time parameters:
    pulse length between 50 and 200 ms
    pause length between two call charge pulses >50 ms.
  - Call charge pulses are no longer transmitted when the exchange receives the idle line criterion from the called line.
- There is no dialling tone supervision.
- In outgoing traffic with register signalling, if a call is answered before dialling is completed, the connection is immediately cleared down.

- If an outgoing exchange call does not receive an answering criterion within 60 seconds of the end-of-dialling signal, the connection is cleared down via forward release. This also applies to "special services".
- A backward release before answering cannot be applied to the line. The line awaits the forward release criterion.
- The blocking criterion can only be applied to the connection from the called line. The blokking criterion is detected after 200 ms. The idle state criterion applied to the line is interpreted as the blocking release signal after 200 ms (outgoing exchange connections are blocked from the exchange side, incoming exchange connections from the PABX side).

Incoming Dialling

Direct dialling-in procedures (DID/DDI) are the same for both register signalling variants (2/5 or 2/6), as long as the corresponding base signal is used.

Outgoing Dialling

For outgoing dialling, the exchange side differentiates between the two register signalling variants. For "2/6" signalling, the entire digit sequence is required up to the end-of-dial signal, whereas for "2/5" register signalling to national destinations, the dialling information is evaluated in max. 2 blocks. The "2/6" variant additionally offers the option of call charge assignment to the station line in the exchange (Tarificaciòn por lìnia TL). This option is set individually for each call and is tranferred to the exchange with the group II signal CLL (Clas de llamada).

The necessary station line identification by the exchange can be carried out either before or after the dialled information is signalled to the exchange. The exchange always requires the entire dialled digit sequence, i.e. area code and subscriber number (Y)ABMCDU.

10.17.5 Audible Tones for Spain

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
1st exchange dial tone	425	Continuous tone
2nd exchange dial tone	600	Continuous tone
Busy tone	425	**200**/200
ATB tone	425	**200**/200/**200**/200/**200**/600

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse/Pause [ms]	Level (dBm0)
Internal dial tone	1	425	**200**/300/**200**/300/**200**/800	-6

Tone	Tsl	Frequency [Hz]	Pulse/Pause [ms]	Level (dBm0)
External dial tone	2	425	Continuous tone	-6
Ringback tone	4	425	**1500**/3000	-6
Busy tone	5	425	**200**/280	-6
Override tone	6	1400	**400**/5000	-28
Call waiting tone	7	425	**600**/200/**600**/1000	-28
Special dial tone	3	425	**1000**/100	-6
NU tone	8	950/1400/ 1800	Triple tone **330-400-330**/1000	-13
Data call tone	9	1300	**600**/1800	-2
2nd exchange dial tone (simulated)	13	600	Continuous tone	-6
Conference auxiliary tone 2	23	1400	Continuous tone	-28
Conference auxiliary tone	24	425	Continuous tone	-28
LCR	26	1800	**340**/200/**340**/200/**340**/1000	- 6

Special tone sequences:

Tone	Sequence	derived from
Conference tone	400 ms 5000 ms 400 ms Continuous	Conference auxiliary tone 2 Pause Conference auxiliary tone 2 Silence
Call waiting tone	Continuous	Call waiting tone
Call waiting tone for night service terminal	Continuous	Call waiting tone
Override tone	Continuous	Override tone

10.17.6 Transfer of the 2nd Trunk Dial Tone

Detection in conformity with the approval regulations of the C.T.N.E.
The 2nd trunk dial tone has the following characteristics:

- Frequency: 600 Hz±5%
- Clock: Continuous signal
- Level: -35 dBm

  The detection time is in the range >1.5 s to <3.5 s. Reception is not affected by:

- AC voltages up to 1 V_{rms} <1s or superimposed voltages at intervals >250 ms in length
- AC voltages less than -45 dBm
- AC voltages less than 160 Hz up to V_{rms}=1 V
- Noise voltages less than 50 ms duration with peaks up to 100 V

The delay up to the 2nd trunk dial tone can be typically 20 s; maximum delay may exceed 60 s (timeout can be set for several minutes).

The digits are transmitted after detection of the dial tone with a delay of at least 0.5 s and no more than 20 s.

The signal is transmitted to the user with both direct listening and a simulated dial tone.
Note:
Direct listening time: several minutes
The dial tone is distinguishable from the internal dial tone and the 1st trunk dial tone (600 Hz!)

The signal is transmitted at the digit location at which the trunk call waits for the 2nd trunk dial tone.

10.17.7 Long Flash Time

Analog station line circuits can be set up for:

- Devices with a flash key for 50 - 130 ms and release detection above 150 ms,
- Devices with a hook flash for 50 - 1000 ms and release detection above 1100 ms.

This stipulation forms part of the C.T.N.E. approval regulations.

A system may contain a mixed configuration of the two device types. A specific SLMA port is assigned statically for one of the two types.

10.18 Mexico

10.18.1 Call Processing Features

Dialled Digit Buffering for Outgoing Connections

For outgoing exchange traffic via **analog lines**, the exchange dialling tone is transmitted to the PABX as an initial seizure acknowledgement. In some cases, the time between initial seizure and the application of the dialling tone by the exchange can take up to 120 seconds.

For this reason, the PABX buffers the dialled digit sequence entered by the PABX user after initial seizure (max. 15 digits), and waits for the exchange dialling tone before transmitting them to the exchange.

In the HiPath system, the maximum delay time between initial seizure of the exchange trunk and the application of the dialling tone by the exchange is set at 30 seconds. If this time is exceeded, the trunk circuit is disconnected and the station user receives a busy signal.

For outgoing exchange traffic via **digital lines,** the exchange transmits a seizure acknowledgement to the HiPath 4000 instead of the dialling tone after initial seizure of an exchange voice channel. Here, too, the dialled digit sequence entered by the PABX user after initial seizure is buffered by the system to compensate for the seizure acknowledgement delay (max. 15 digits). As for analog traffic, the exchange connection is terminated after timeout, and the user receives a busy tone; the maximum permissible delay is set in the HiPath 4000 to 2 seconds.

Attendant Intercept of DID (DDI) Exchange Connections

Direct-dialled incoming exchange calls are re-rung at the attendant console as attendant intercept calls if the dialled PABX user does not answer within a given time.
The answer timer can be set via AMO.

10.18.2 Bothway Exchange Traffic with MOSIG (HKZ) Signalling (TMANI)

2-wire a/b exchange lines with HKZ (MOSIG) signalling

Feed voltage from exchange:48V +/- 10% with 2 x 400 ohms

No direct inward dialling

Lines can be configured for bothway and one-way traffic

Dialling information signalling:pulse signalling and DTMF signalling

The exchange does not supply call charge pulses.

The following criteria are detected in incoming traffic:

- Seizure (with AC ringing voltage)
  - Frequency: 25 Hz +/- 5 Hz

  - Voltage: 90 V_{rms} +/- 5%
  - Make/break: 1000 / 4000 ms
- Forward release (exchange feed voltage interrupt)

The following criteria are signalled in incoming traffic:

- Answering criterion (loop closure in the PABX)
- backward release (by opening PABX loop)

The following criteria are detected in outgoing traffic:

- exchange dialling tone as seizure acknowledgement
- backward release (exchange feed voltage interrupt)

The following criteria are signalled in outgoing traffic:

- initial seizure (loop closure in the PABX)
- DTMF or pulse dialling signals
- forward release (by opening PABX loop)

The following features are not implemented:

- Re-answering
- Calibrated interruption (flash to exchange)
- Polarity inversion
- Time limit (coin telephone)

Loop impedance (including PABX line circuit) must be less than 1800 ohms.

Leak resistance between the a and b-wires or a/b-to-ground leak resistance must be greater than 20 kohms.

Exchange dialling tone detection
The maximum delay between seizure and dialling tone detection is 30 s (default value).

10.18.3 Bothway Tie-Trunk Traffic with E&M Signalling (TMEW2)

4-wire lines for voice transmission
(2 wires for "mouth" transmission + 2 wires for "ear" transmission)

2-wire connections for E&M signalling transmissions

Dialling information signalling:pulse signalling and DTMF signalling

Traffic direction:bothway

10.18.4 Incoming Exchange Traffic with DID (DDI) via DIUN2

4-wire 2 Mbit/s connections via PCM30 highways with channel-associated signalling

2-bit line signalling ('a'-bit and 'b'-bit) via HDLC channel 16 as per ITU Q.421, Q.422 and Q.424.
The signalling criteria are sent in continuous signal mode and are identical for incoming and outgoing traffic:

- Idle state
- seizure
- seizure acknowledgement
- answering criterion
- clear back criterion
- connection release
- release acknowledgement
- backward blocking criterion

Register signalling via MFC-R2 registers as per ITU Q.440 to Q.480 via voice channels with forced frequency passage.

Each of the 30 B-channels of a PCM30 exchange highway can be separately configured for one-way traffic (incoming or outgoing) as desired.

Direct dialling-in (DDI/DID) only.

The PABX always receives the last 4 digits of the dialled sequence from the exchange.

The exchange does not supply call charge pulses.

In outgoing traffic, the exchange transmits a seizure acknowledgement to the PABX instead of the exchange dialling tone.

Notes on incoming traffic

- In "linked call transfers", repeated attendant intercept re-rings may occur. The clear back criterion is not transmitted to the exchange each time an internal user goes on-hook.
- No release request signal

- No release monitoring after clear back signal is transmitted to the exchange
- HiPath 4000 does not require rapid release function via answering criterion during register signalling.
- Backward release by the line circuit before answering is not possible.

Notes on outgoing traffic

- If no seizure acknowledgement is received within 2 seconds after seizure, HiPath applies a busy tone to the user's line. The forward release to the exchange is not transmitted until the seizure acknowledgement signal is received.
- No exchange dialling tone detection. User receives simulated dialling tone on receipt of exchange seizure acknowledgement.
- If the clear back signal is applied for more than 600 ms following a forward release signal, this is interpreted as an incoming blocking signal.
  In all other cases, the release acknowledgement signal is awaited (no timer supervision).
- If a clear back signal is received, a forward release is initiated.
- Following a connection release and the connection release acknowledgement an outgoing re-seizure delay timer of 250 ms is started (default value).
- In the case of 'c' and 'd'-bit errors ('c' = 0, 'd' = 1), bits 'a' and 'b' are not evaluated.
  A line alarm is signalled after 4 seconds.
  Following the line alarm, the line is monitored for a further 4 seconds.
- Line blocking is detected after 2 seconds.
  The blocking release signal is the idle line criterion, which is applied during the re-seizure delay time. Incoming seizures during this time are evaluated.

10.18.5 Digital Tie-Trunk Traffic

Telmex (Telefonos de Mexico) offers leased lines for digital networked systems. The signalling criteria are defined by the networked PABX operators. The PCM30 highways comply with the ITU recommendations G.703 / 704 / 821.

10.18.5.1 Bothway Tie-Traffic with E&M Signalling (DIUN2)

Digital network links via DIU-CAS with E&M signalling and pulse dialling or DTMF dialling on station lines (for mixed node networks, i.e. HiPath and foreign vendor products).

Signalling is carried out via the 'a'-bit of the HDLC signalling channel (channel 16).
Functionality as for E&M analog network links in V3.1

10.18.5.2 Bothway Tie-Traffic via S2 Lines with DPNSS1 Signalling (DIUT2)

Digital network links via S2 lines to DIUT2 and CDG for DPNSS1 signal conversion (for connecting EMS601/OMNI systems), with HiPath ISDN features (basic call setup + supplementary features).

10.18.5.3 Bothway Tie-Traffic via S_0/S_2 Lines with CorNet-NQ Signalling (STMD/DIUT2)

Digital network links via S0/S2 lines with CorNet-NQ signalling (for networking with other HiPath systems)

10.18.6 Audible Tones for Mexico

The following audible tones are to be recognised by the PABX::

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone	425	Continuous tone
Ringback tone	425	**1000**/4000
Busy tone	425	**250**/250
Overload (ATB tone)	425	**250**/250

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse/Pause [ms]	Level (dBm0)
Internal dial tone	1	425	Continuous tone	-10
External dial tone	2			
Special dial tone	3	425	**1400**/100	-10
Ringback tone	4	425	**1000**/4000	-10
Busy tone	5	425	**250**/250	-10
Override tone	6	1400	**200**/2000	-13,5
Call waiting tone	7	425	**160**/140/**160**/4000	-15
NU tone	8	950/1400/ 1800	Triple tone **340-340-340**/1000	-10
Data call tone	9	1300	**600**/1800	-1

Tone	Tsl	Frequency [Hz]	Pulse/Pause [ms]	Level (dBm0)
Conference auxiliary tone 2	23	1400	Continuous tone	-13,5
Conference auxiliary tone	24	425	Continuous tone	-15
LCR	26	1800	**340**/200/**340**/200/**340**/1000	- 10

Special tone sequences:

Tone	Sequence	derived from
Conference tone	200 ms 2000 ms 200 ms Continuous	Conference auxiliary tone 2 Pause Conference auxiliary tone 2 Silence
Call waiting tone	Continuous	Call waiting tone
Call waiting tone for night service terminal	Continuous	Call waiting tone
Override tone	Continuous	Override tone

10.19 Poland

10.19.1 Call Processing Features

Detecting First and Second Exchange Dialling Tones for Outgoing Calls

For MFC-R2 register signalling in outgoing traffic via digital exchange lines, the exchange transmits the seizure acknowledgement signal to the PABX instead of the exchange dialling tone when the voice channel of the exchange line is seized. The dialled digit sequence is then transmitted to the exchange by the PABX.

For pulse dialling in outgoing traffic the exchange applies the first exchange dialling tone after sending the seizure acknowledgement criterion. Only then is the dialled digit sequence transmitted to the exchange.

In pulse dialling, depending on the dialled digit sequence, i.e. 8xx... or 8xxx...., a second dialling tone is applied.

The remaining digits in the dialled sequence in these cases are not transmitted to the exchange until the second dialling tone has been received. The delay between initial seizure and the detection of the seizure acknowledgement from the exchange can take up to 2 seconds. If this time is exceeded, the exchange circuit is disconnected and the PABX user receives the busy tone.

The exchange dialling tone is also evaluated for speed-dialling and saved number re-dialling. However, the exchange tone is only passed on to the PABX user if he or she dials manually.
For MFC-R2 register signalling, the second exchange dialling tone is not transmitted to the PABX.

10.19.2 Analog Exchange Traffic

Bothway Exchange Traffic with MOSIG (HKZ) Signalling (TMANI)

The following features are not implemented:

- outgoing traffic:
  – no answering criterion or clear back criterion detection
  – no idle line monitoring
  – no register re-call signals (B-signal to exchange)
- incoming traffic
  – no register re-call signals

Incoming Exchange Traffic with DID (DDI) via TMLSF

The following features are not implemented:

- Override / intrusion function for public network operator

10.19.3 Bothway Exchange Traffic via DIUN2

Digital exchange connections with thefollowing features:

- 4-wire 2 Mbit/s connections via PCM30 highways with channel-associated signalling.
- 2-bit line signalling ('a'-bit and 'b'-bit) via HDLC channel 16 as per ITU Q.421, Q.422 and Q.424.
  The signalling criteria are sent in continuous signal mode and are identical for incoming and outgoing traffic, with the exception of pulse dialling signals, call charge pulses and forced release.
  The exchange only transmits call charge pulses to the PABX for the "Metering Variant".
- Register signalling via MFC-R2 registers as per ITU Q.440 to Q.480 via voice channels with forced frequency passage.
  The called line identification information is transmitted according to the Polish R2 signalling variant, which represents a sub-group of the various signals specified by the ITU recommendations.

Dialling information signalling:

- Pulse dialling
  - Incoming traffic, applies to basic variant:
    Frequency: 10 ± 2 Hz
    Pulse/pause: 2 ± 50 %
    Inter-dial pause: > 400 msec
  - Outgoing traffic, applies to basic and metering variant:
    Frequency: 10 ± 0,5 Hz
    Pulse/pause: 66/33
    Inter-dial pause: ≥ 800 msec
- MFC-R2 (incoming / outgoing traffic, applies to basic and metering variants)
  - Register type: 2 of 6 per ITU Q.440
  - Frequency levels: 8dBm0 ±1 dB
- Each of the 30 B-channels of a PCM30 exchange highway can be separately configured for one-way traffic (incoming or outgoing) as desired.
- Direct dialling-in (DDI/DID) only
- Two optional signalling variants can be configured:
  - Incoming and outgoing traffic without exchange call charge pulse (basic variant)
  - Outgoing traffic only, with exchange call charge pulse (metering variant)
- In outgoing traffic, the exchange transmits a seizure acknowledgement to the PABX instead of the exchange dialling tone.
- In pulse dialling, depending on the dialled digit sequence, i.e. 8xx... or 8xxx...., a second exchange dialling tone is applied.

10.19.4 Analog Tie-Trunk Traffic

10.19.4.1 Pulse Dialling, Bothway traffic with E&M Signalling (TMEW2)

Carrier frequency signalling,
E&M pulse signalling via 4-wire lines.
No known restrictions.

Carrier frequency signalling, E&M pulse signalling via 2-wire lines
Since no 2-wire E&M pulse signalling lines exist in the HiPath, the existing 4-wire lines (TMEMW, TMEW2 from HiPath 4000) are connected to an external hook-switch which transfers the signals to the Polish 2-wire lines.
No known restrictions.

10.19.4.2 Loop-Start Signalling via TMANI and SLMA

Out-dialling in loop-start mode

- HW: TMANI
- The associated DH 120 VA cannot execute dynamic level-switching for transit traffic. For transit connections via these lines (tie-traffic in transit), some echoing must be taken into account. In order to minimise this effect, only the level for short lines is switched for these paths, regardless of the actual call path length.

In-dialling in loop-start mode

- HW: SLMA
  For loop-start signalling, adjustments to the input and output levels might be necessary. These would be in the form of new SICOFI parameters for the loadware variant, which can be implemented at short notice if required.

10.19.4.3 Local Battery Signalling

A basic signalling protocol for local battery signalling is in progress.

HW:TMBCT with 25/50-Hz ringing generator.

User terminals and line repeaters with local battery signalling can be connected as long as they can provide the signalling criteria 'seizure' and either 'release' or 'release acknowledgement'.
Further development is necessary for other signalling protocols.

10.19.5 Audible Tones for Poland

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone	425	Continuous tone
Ringback tone	425	**1000**/4000
Busy tone	425	**500**/500

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	**Pulse**/Pause [ms]	Level (dBm0)
Internal dial tone	1	425	Continuous tone	-10
External dial tone	2			
Special dial tone	3	425	**1400**/100	-10
Ringback tone	4	425	**1000**/4000	-10
Busy tone	5	425	**500**/500	-10
Override tone (warning tone)	6	425	**100**/4900	-12
Call waiting tone (call waiting tone)	7	425	**160**/140/**160**/4000	-12
NU tone	8	950/1400/ 1800	Triple tone **340-340-340**/1000	-10
Data call tone				
2nd exchange dial tone (simulated)	13	425+ 330	Continuous tone	-10 -9
Conference auxiliary tone	24	425	Continuous tone	-12
LCR	26	1800	**340**/200/**340**/200/**340**/1000	- 10

Special tone sequences:

Tone	Sequence	derived from
Conference tone	Continuous	Pause
Call waiting tone	Continuous	Call waiting tone
Call waiting tone for night service terminal	Continuous	Call waiting tone
Override tone	Continuous	Override tone

10.20 Czech Republic

10.20.1 Call Processing Features

Outgoing Line Seizure with MFC-R2 Register Signalling

If MFC-R2 register dialling is used to the exchange, a minimum digit sequence composed of a route discrimination code, a local exchange code, area code or international code may be required before an ougoing line can be seized. This is due to the various MFC-R2 register dialling stage systems in the public network (PK202, PK22).

Camp-On/Knocking (Call Waiting Indication)

Czech requirements dictate that call waiting indications must be prioritised, i.e. the call waiting (knocking) signal must depend on the priority (= importance) of the calling party. Priority signalling is currently not implemented in the HiPath 4000 for this feature. For this reason, call waiting indication via exchange lines (including the "automatic" operator variant) is inhibited. This does not affect the internal feature for HiPath users, which is still offered.

10.20.2 Exchange Traffic

The exchange traffic signalling adaption for Czech public network exchanges is carried out in HiPath 4000 exclusively by means of DIUN2 signalling protocol conversion, i.e. all exchange signals to both analog and digital exchanges are transmitted via the digital exchange lines of the DIUN2. For signal conversion from digital to analog and vice-versa is carried out with KZU signal converters.

In accordance with Czech requirements, signalling is carried out over 4-wire PCM highways according to the ITU recommendations G703, G704 and G732 with channel-associated signalling, and via the 'a' and 'b'-bits of the HDLC signalling channel (channel 16).

MFC-R2 register signalling

The Czech MFC-R2 register signalling variants is based on the ITU recommendations Q.440 - Q.480 with some additional national modifications.

The delay time between a backward signal with forced passage and the initiation of a pulse signal is t> 300 ms.

The parameters for MFC-R2 dialling information signalling correspond to those defined in the ITU recommendations Q.454 and Q.455.

- Frequencies for forward signals:
  - f0 = 1380 Hz
    f1 = 1500 Hz
    f2 = 1620 Hz
    f3 = 1740 Hz
    f4 = 1860 Hz
    f5 = 1980 Hz
- Frequencies for backward signals:
  - f0 = 1140 Hz
    f1 = 1020 Hz
    f2 = 900 Hz
    f3 = 780 Hz
    f4 = 660 Hz
    f5 = 540 Hz
- Transmitter parameters:

- Nominal frequency deviation: ± 4 Hz
- Absolute power level per non-modulated signalling frequency: -8 dBm0 ±1 dB
- Max. deviation of both frequencies: < 1 dB

• Receiver parameters:

- Guaranteed range of reception: -31 dBm0 to -5 dBm0

The signals for standard connection release and release after timeout correspond to ITU Q.475 and Q.476:

• The transmission monitoring time for a seized outgoing register for transmitting a signal is:
12 s≤T1≤ 18 s (transmission time)

• The transmission monitoring time for a seized outgoing register between two signals is:
25 s≤ T2 ≤ 30 s (idle time)

• The receive monitoring time for a seized incoming register for signal detection is:
15 s≤T3≤ 24 s

If the timer of the seized outgoing register runs out, the exchange connection is terminated. The PABX user receives a 'special information tone'.

If the timer of the seized incoming register runs out, the A4 signal is transmitted, and the PABX awaits the forward release signal from the exchange.

Pulse dialling

Pulse dialling is possible for incoming and outgoing dialling to all exchange types, with the exception of analog exchanges with U-interface signalling via KZU signal converters, which do not support direct inward dialling.

• Incoming exchange traffic:

- Frequency: 8 ... 14 pulses/s
Pulse length: 15 ... 100 ms
Pause length: 15 ... 75 ms
Inter-dialling pause: 150 ms <t<10 s

• Outgoing exchange traffic:

- Frequency: 10 Hz
Pulse length: 50 ms ± 10 %
Pause length: 50 ms ± 10 %
Inter-dialling pause: 700 ... 800 ms
Exchange dialling delay: 100 ... 800 ms

DTMF dialling

- Incoming traffic: Not planned
- Outgoing traffic: According to ITU Q.23
  - Smallest signal pair length defineable in the HiPath
    Pulse/pause: 80/80 ms
  - Lower frequencies: -7,2 dBm0
  - Higher frequencies: -5,2 dBm0
  - Tolerance per frequency: ± 1,8 %

10.20.3 Digital Exchange Traffic

10.20.3.1 Digital Exchange

Signalling for one-way and bothway traffic.

Features for incoming traffic

- Dialling information signalling:MFC-R2 or pulse dialling
- User answering supervision
- After 180 seconds, attendant intercept or all-trunks-busy signal until caller goes on-hook. In HiPath 4000HiPath 4000, either attendant intercept after timeout or ringing tone until forward release is received from exchange (AMO-defined).
- Following backward release (by PABX user), the all-trunks-busy signal is switched to the line, and the clear back signal sent to the exchange. The PABX awaits a forward release signal from the exchange in return.
- Immediate busy tone to caller or attendant intercept if called party is busy. Override (intrusion) signal from the digital exchange leads to attendant intercept with immediate ringing tone to the exchange line, or to override/ intrusion on called party's connection (three-way conference with override notification tone) and override tone to exchange line.
- If the PABX attendant answers while the public exchange operator is still pressing the override key, the answering signal is not sent to the exchange until the override tone is finished.
- No forced release
- Exchange operator re-ring (optional), PABX transmits "11" for 100 ms.
- The audible tones of the HiPath system are switched without timer-dependencies

- Malicious call trace
  HiPath 4000 users must be explicitly assigned the malicious call trace classmark in order to initiate a call trace request.
  The transmission of the malicious call trace request signal back to the calling party's system depends on whether the call is chargeable or not (depends on called party) and whether the call was received via an exchange line with MFC-R2 or pulse dialling information signalling.
  If the malicious call trace request is not followed up by a confirmation signal before the called party goes on-hook, the clear back signal is sent to the source system and the connection is released.
  If the called party has confirmed the malicious call trace request (via a dialled code), a print-out is started on the service terminal printer with the following data:
  - Date and time of call
  - Station number of trace initiator
  - Port equipment number of exchange line concerned
  - Calling line identification number if available (MFC-R2 only)
    In this state, the connection can no longer be cleared down by the called PABX user, the clear back signal is not transmitted to the exchange. The connection can only be cleared manually via the central attendant console, by means of blocking and unblocking the line.
  - Forward release from the exchange has priority over malicious call trace requests and confirmation from the PABX.

Signalling criteria of the incoming exchange registers of the HiPath 4000

- Idle line signal
- seizure
- seizure acknowledgement
- dialling information
- override
- answering
- operator re-ring
- malicious call trace confirmation
- clear back
- forward release
- blocking

Features for outgoing traffic

- Dialling information signalling:MFC-R2, pulse dialling and DTMF
- No R2 backward signal for switching to pulse signalling
- No exchange dialling tone detection for R2 register dialling, but optional second exchange tone detection for pulse dialling and DTMF dialling.
- Signalling variants
  - Metering
  - Universal

toll calls:	withoput malicious call trace with malicious call trace
local calls:	answering or malicious call trace or answering with malicious call trace or answering with clear back signal.

- Special services in the exchange can be activated by means of flash pulse "01" and via DTMF dialling.
- In outgoing exchange dialling, a minimum digit sequence (between 3 and 22 digits) must be dialled, depending on the destination, before an exchange line is seized. In these cases, no simulated exchange dialling tone is sent to the PABX user.
- Malicious call trace
  If a HiPath 4000 user with the malicious call trace feature is called, a malicious call trace request (= request for calling line data) is automatically set up to the caller's system.
  When the caller goes on-hook at the end of the call, the forward release criterion is applied to the line with a 30-second delay. If the called party hangs up during this time, no print-out is started. If not, a print-out is started as for malicious call trace in incoming traffic, with the addition of the caller's station number.
  Variants:
  If the called party does not hang up within 30 s, one of the following two possibilities may occur:
  - Print-out and forward release
  - Print-out without forward release. In this case, the connection can only be cleared back via the attendant console or from the exchange.

Signalling criteria of the outgoing exchange registers of the HiPath 4000

- Idle line signal
- seizure
- seizure acknowledgement

- dialling information
- answering
- flash signal
- call charge pulse detection
- backward release
- forced release
- forward release
- malicious call trace acknowledgement
- blocking

10.20.3.2 Analog Exchange with P-Interface and Signal Converter

The same conditions apply as for digital signalling without KZUs. Whether the local public network exchange is a digital exchange or an analog exchange with P-interface is irrelevant for the HiPath in incoming exchange traffic, since the KZU converts the signalling information from the analog exchange to a digital signalling protocol.

10.20.3.3 Analog Exchange with U-Interface and Signal Converter

Signalling for one-way and bothway traffic. Exchange seizures have priority over initial seizures of PABX users.

Features for incoming traffic

- No malicious call trace, since the U-interface is routed via a KZU.
- Ringing voltage detection by the KZU. The KZU transmits the seizure signal and not the ringing cadence.
- Ringing interrupt monitoring in the KZU over 12 s
- If no new ringing signal is detected by the KZU within these 12 seconds, the KZU sends a clear forward signal to the HiPath 4000.
- Flash signal from HiPath to exchange is possible.
  The KZU transmits a clear back signal of less than 300 ms directly to the exchange.
  Only if the signal is longer than 300 ms will it be interpreted as a connection release.
  The HiPath sends a flash signal of 100 ms to the KZU.

- Use of "special service" feature by called party. The called party can access the special services in the exchange by dialling a flash signal followed by a DTMF signal code. The Czech Telcom defines which special services can be accessed. DTMF backward signalling is allowed in this case.
- Blocking signal to KZU and vice versa ('a'-bit = 1, 'b'-bit = 1)
- The answering signal is transmitted to the exchange by the KZU after 1st ringing pulse at the earliest
- Forward release during answer signal transmission is acknowledged with 11, 10

Signalling criteria of the incoming exchange registers of the HiPath 4000HiPath 4000

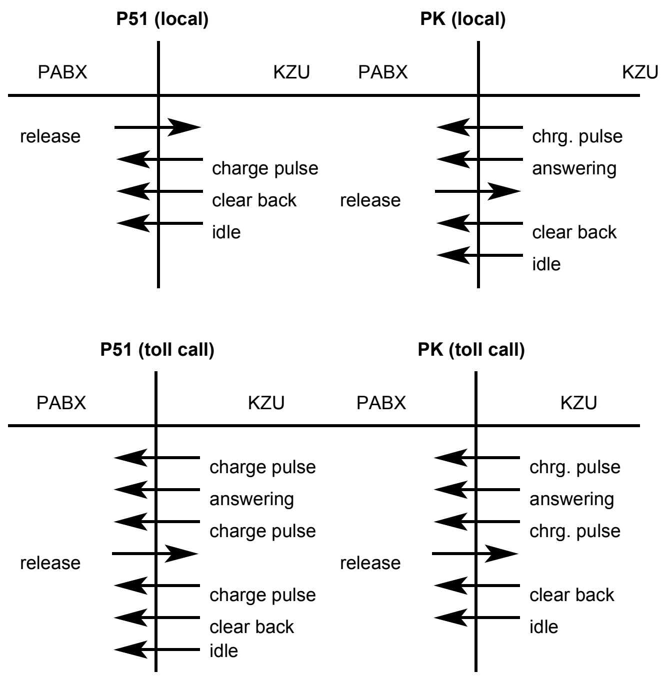
- idle line signal
- seizure
- seizure acknowledgement
- answering
- clear back
- forward release before answering
- forward release during answering signal transmission
- flash to exchange
- blocking

Features for outgoing traffic

- Dialling information signalling: MFC-R2, pulse dialling
- Forward release only
- No idle monitoring in KZU
- Flash to exchange (100 ms)
- Use of "special service" feature by called party. The called party can access the special services in the exchange by dialling a flash signal followed by a DTMF signal code. The Czech Telcom defines which special services can be accessed. DTMF backward signalling is allowed in this case.
- Blocking signal to KZU and vice versa ('a'-bit = 1, 'b'-bit = 1)
- The signal combinations for call charge pulse transmission and the release criterion depend on
  – local exchange type (P51 or PK) to which the HiPath 4000 is connected and

– whether the call is a local call or a long-distance call.

For all variants, the answering criterion is simulated in the KZU when the first call charge pulse is received (bit sequence 01) and transmitted to the HiPath if required.



- Dial tone monitoring for 1st exchange dialling tone (425 Hz, Morse a) for DTMF and pulse dialling (optional) information signalling. Monitoring time max. 5 seconds.
- Start of out-dialling ≥ 500 ms after dialling tone is detected.
- No second exchange dialling tone
- No malicious call tracing, since the U-interface is routed via a KZU.
- Suffix dialling possible, if HiPath 4000 user has set up the call with a DTMF telephone
- No call charge pulse simulation
- End-of-dialling and answering criterion simulation in the system

Signalling criteria of the outgoing exchange registers of the HiPath 4000

- idle line signal
- seizure

- seizure acknowledgement
- dialling information
- answering
- flash signal
- call charge pulse detection
- call charge pulse detection during forward release
- forward release
- blocking

10.20.4 Incoming Exchange Traffic with DID (DDI) via TMAG

3-wire exchange line for DC signalling system.

Exchange line module:TMAG with LW variant PZGTMAG1

Line adapter S97810-Q3011-X.* with resistance > 900 ohms on b-wire, loop current < 90 mA.

Connections as specified in Telcom specification "Technical Specification of PABX Interfaces in Czechoslovak Telecommunication Network".

Dialling information signalling:pulse dialling.

10.20.5 Audible Tones for the Czech Republic

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone	425	**330**/330/**660**/600
Ringback tone	425	**1000**/4000
Busy tone	425	**320**/340
Overload (ATB tone)	425	**120**/120

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse/Pause [ms]	Level (dBm0)
Internal dial tone	1	425	Continuous tone	-5
External dial tone	2	425	**320**/340/**660**/600	- 5

Tone	Tsl	Frequency [Hz]	**Pulse/**Pause [ms]	Level (dBm0)
Special dial tone also ATB tone	3	425	**160**/160	-5
Ringback tone	4	425	**1000**/4000	-5
Busy tone	5	425	**320**/340	-5
Override tone	6	425	**320**/340/**320**/1500	-11
Call waiting tone 1	7	425	**320**/5000	-11
NU tone	8	950/1400/ 1800	Triple tone **320-320-320**/1000	-5
Data call tone				
ATB tone	11			
Call waiting tone 2	15	425	**1000**/160/**340**/3500	-5
Conference auxiliary tone 2				
Conference auxiliary tone	24	425	Continuous tone	-11
LCR	26	1800	**340**/200/**340**/200/**340**/1000	- 5

Special tone sequences:

Tone	Sequence	derived from
Conference tone	600 ms	Internal dial tone
Call waiting tone	Continuous	Call waiting tone 1
Call waiting tone for night service terminal	Continuous	Call waiting tone 1
Override tone	Continuous	Override tone

10.21 Denmark

10.21.1 Call Processing Features

10.21.1.1 Operator Re-Ring for Incoming Exchange Calls

This feature is implemented in two different solutions peculiar to the Danish telecommunication network as follows:

- If a called PABX user does not answer an incoming exchange call, the call is transferred to the PABX attendant with an attendant re-ring status signal. This re-ring line signal is only transmitted for incoming direct-dialled

exchange calls via MFC-R2 register signalling.
If attendant intercept after timeout is implemented for incoming exchange calls, the attendant re-ring signal can be ignored (AMO-definable).

- If a called PABX user terminates an incoming exchange call (i.e. goes on-hook), the exchange call is not cleared back by the exchange. Instead, the exchange sends an "operator re-ring" signal to the PABX, which immediately rings through to the attendant console.
  The time between the called PABX user going on-hook and the transmission of the "operator re-ring" signal by the exchange is max. 120 seconds (90 ± 30 s).
  This operator re-ring is possible for incoming direct-dialled exchange calls via MFC-R2 register signalling and can be repeated as often as desired.

10.21.1.2 Digit Analysis and Digit Transfer for Outgoing Calls

For outgoing traffic, the dialled digit sequence is transmitted to the exchange by means of DTMF signals via lines with P1 signalling criteria.

The dialled digit sequence is evaluated as follows:

- First dialled digit
  - The exchange trunk access code is always a "0".
  - The "9" is always the attendant console access code
  - The number 1 ... 8 is always the first digit of an internal station number
- Long-distance calls (toll calls)
  The Danish public network has an 8-digit numbering plan. Only special services (e.g. emergency services) in the public network are reached via 3 - 4-digit codes.
- International calls
  International call numbers differ in length. The route discrimination code for international calls (international dialling code) is "09" or "009" for a PABX user, i.e. the HiPath 4000 system evaluates the 3rd digit of a dialled digit sequence in order to determine international calls.
- External toll/code restriction
  The dialled digit sequence following a leading "0" (exchange access code) is checked against the external toll/code restriction tables (EWAKO) in theHiPath 4000 to determine whether the user is allowed to set up the call.
- Voice connection setup
  The PABX sets up the voice connection between the user and the exchange line after digit analysis of the initial digits, whether the end-of-dial signal has

been transmitted by the station line or not. If the end-of-dial signal is transmitted before the digit analysis has been completed, the connection is set up immediately.

10.21.1.3 Suffix Dialling on Outgoing Exchange Connections

DTMF signals can be transmitted from the HiPath 4000 to the exchange on an open line, e.g. in order to dial a user in a remote PABX with quasi-DID via the exchange. The DTMF signal transmission can be initiated at any digital or analog voice terminal in the HiPath system.

10.21.1.4 Local Announcements for Incoming Exchange Calls

For incoming exchange traffic to the attendant of a PABX, calls can be switched via an announcement unit. The announcement is heard by both the exchange caller and the PABX attendant, and is intended for standard announcements such as company name and/or location. The announcements are configured separately for each PABX via AMO.

10.21.1.5 Incoming Line Seizure with Quasi-Inward Dialling

On receiving the incoming seizure signal, the PABX can

- switch a DTMF receiver to the line,
- apply a dialling tone to the line and
- transmit the answering signal to the exchange

as an alternative to switching a call to the attendant console.

This offers external users with DTMF telephones the possibility of direct dialling-in to a PABX user via the speech channel of the incoming connection.

A delay of at least 500 ms between switching the DTMF receiver across the line and sending the answering signal to the exchange ensures that the last digit of the number dialled by the external caller is not mistakenly evaluated as the first digit of the PABX user's station number.

The PABX user's station number in this case is a 3 or 4-digit code. Special numbers, e.g. the attendant console access code, are single-digit codes.

If no DTMF signals are received within <10 seconds of sending the dialling tone to the exchange, the connection is transferred to the attendant. The same applies if the DTMF signal combination is incomplete.

An end-of-dialling signal is not sent by the exchange.

10.21.2 Exchange Traffic

10.21.2.1 Analog Exchange Traffic

Bothway exchange traffic (MOSIG) with 16-kHz call charge pulse (TMANI).

10.21.2.2 Digital Exchange Traffic

In HiPath 4000, the digital exchange traffic connection to the Danish public network is implemented via PCM30 highways with channel-associated signalling for MFC-R2 incoming traffic and DTMF outgoing traffic.

Three different line signalling criteria systems are implemented for 3 basic exchange connection variants:

1. Incoming traffic
   - via attendant
   - with quasi-DID
     - Instead of switching a call to the attendant console, the PABX can switch a DTMF receiver to the line,
       apply a dialling tone to the line and
       transmit the answering signal to the exchange.
     - A delay of at least 500 ms between switching the DTMF receiver across the line and sending the answering signal to the exchange ensures that the last digit of the number dialled by the external caller is not mistakenly evaluated as the first digit of the PABX user's station number.

   The **line signalling criteria (1)** for these call types consist of :
   - idle line
   - seizure /ringing
   - answering
   - forward release
   - backward release
   - operator re-ring
   - blocking
2. Incoming exchange traffic with DID/DDI
   The **line signalling criteria (2)** for this call type consist of :

- idle line
- seizure
- seizure acknowledgement
- R2 register signalling
- end-of-dial signal
- answering
- forward release
- backward release
- operator re-ring
- blocking

3. Outgoing traffic
   The **line signalling criteria (3)** for this call type consist of :
   - idle line
   - seizure
   - seizure acknowledgement
   - dialling information, DTMF
   - end-of-dial signal (optional)
   - answering (optional)
   - talking state
     – without answering
     – with answering
   - called party goes on-hook (only possible from talking state with answering)
   - forward release
   - blocking
   - backward release
   - call charge pulses for
     – lines without answering
     – lines with answering
4. The individual lines can be operated as follows:
   - incoming exchange traffic with attendant intercept

    - line signalling criteria (1)
- incoming exchange traffic with quasi-DID via DTMF
    - line signalling criteria (1)
- incoming exchange traffic with DID via MFC-R2
    - line signalling criteria (2)
- outgoing exchange traffic with DTMF dialling
    - line signalling criteria (3)
- bothway exchange traffic with attendant intercept and DTMF dialling
    - line signalling criteria (1 and 3)
- bothway exchange traffic with quasi-DID via DTMF
    - line signalling criteria (1and 3)
- bothway exchange traffic with DID via MFC-R2
    - line signalling criteria (2 and 3)

5. Exchange dialling tone detection
   The dialled digit sequence is not transmitted by the PABX until the exchange dialling tone has been received (425 Hz continuous tone).
   In some cases, the delay between initial seizure of an exchange line and dialling tone detection by the PABX can take up to 10 s.
   The HiPath 4000 system applies a simulated exchange tone to a PABX user's line as soon as a free exchange trunk circuit is seized.
   In order for the system to differentiate between the exchange dialling tone and the busy tone, the exchange dialling tone must be applied for at least 500 ms without interruption.
6. Dialling information signalling
   - DTMF out-dialling
       - in outgoing traffic via line signalling criteria (3)
       - Frequencies:
         2 out of 4 (16 tones) as per ITU Q.23
       - transmission level
         for lower frequencies: $\geq$-25$\leq$-4 dBm0
         transmission level for the higher frequencies: $\geq$-25$\leq$-4 dBm0
         high/low frequency difference <5dB
   - transmission cadence:80 ms tone / 80 ms pause
   - DTMF receiving

  – receiver sensitivity in incoming traffic with line signalling criteria (1) and quasi-DID as per ITU Q.23

- MFC-R2 in incoming traffic with line signallling criteria (2)
  – transmission level for MFC signals: ≥-29,5≤-3,5 dBm0

10.21.3 Audible Tones for Denmark

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone	425	Continuous tone
Ringback tone	425	**1000**/4000
Busy tone	425	**240**/240
ATB tone	425	**500**/500

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse/Pause [ms]	Level (dBm0)
Internal dial tone	1	400+450	Continuous tone	-6.5
External dial tone	2	425	Continuous tone	- 6.5
Special dial tone	3	480+620	**20**/20	-6.5
Ringback tone	4	425	**1000**/4000	-6.5
Busy tone	5	425	**240**/240	-6.5
Override tone	6	425	**20**/500	-6.5
Call waiting tone	7	425	**200**/200/**200**/3600	-12.5
NU tone	8	950/1400/ 1800	Triple tone **320-320-320**/1000	-6.5
Positive acknowledgement tone	10	425	**320**/20	-6.5
ATB tone	11	425	**500**/500	-6.5
Conference auxiliary tone 2				
Conference auxiliary tone	24	425	Continuous tone	-12.5
LCR	26	1800	**340**/200/**340**/200/**340**/1000	- 6.5

Special tone sequences:

Tone	Sequence	derived from
Conference tone	20 ms 500 ms 20 ms 500 ms 20 ms Continuous	External dial tone Pause External dial tone Pause External dial tone Silence
Call waiting tone	Continuous	Call waiting tone
Call waiting tone for night service terminal	Continuous	Call waiting tone
Override tone	Continuous	Override tone

10.22 Brazil

10.22.1 Call Processing Features

10.22.1.1 Operator Re-Ring for Incoming Exchange Calls

The re-ring is initiated by the remote exchange operator, when a called, busy PABX user goes on-hook. The feature is only used for specific projects. The necessary signalling is contained in the signalling criteria plans for analog DID exchange trunks (DID) and digital exchange trunks with E&M signalling.

10.22.1.2 Dialling Information Transmission for Outgoing Exchange Calls

In outgoing exchange traffic via digital exchange lines with R2 signalling, the exchange requires a calling line identification number in order to assign the call charges. In this case, the PABX must either transmit the true number consisting of station number + PABX number + local area code (up to 10 digits) or a virtual number assigned in the PABX for all PABX users. This virtual station number can also have up to 10 digits and is AMO-defined.

10.22.1.3 Collect Calls

In Brazil, subscribers can set up collect calls by dialling a special collect call code in the exchange. In order to prevent misuse, the called party hears a collect call notification (cannot be evaluated by the system), and can decide whether or not to hang up within a specific time without being assigned the call charges.

In the HiPath 4000 system, a special collect call authorization or classmark must be set in the user's COS for accepting collect calls.

All calls to users without this classmark are acknowledged by the HiPath 4000 system by sending a short backward release signal to the exchange followed immediately by an answering criterion. This ensures that collect calls via the exchange are cleared back, whereas standard incoming exchange connections remain set up.

10.22.1.4 Outgoing Exchange Connection Setup to Non-Chargeable Destinations

Connections to certain destinations, e.g. special services announcements, are free-of-charge in Brazil. Since the exchange does not transmit call charge pulses to the PABX, these destinations must be entered in the dial plan of the HiPath 4000 system so that they can be detected by digit analysis.

In analog exchange traffic with MOSIG (HKZ) signalling, the exchange also omits to send an answering criterion for non-chargeable destinations. In these cases, the HiPath 4000 system proceeds as follows:

- Outgoing MOSIG (HKZ) traffic with silent reversal on answer:
  In addition to the "real" answering criterion, the HiPath 4000 system generates the same timer-dependent simulated answering criterion as for HKZ signalling without silent reversal. If a "real" answering criterion is detected before timeout, this is evaluated as the connect signal (start record) for the call data recording system.
  The call data system only accepts whichever criterion it is sent first, whether simulated or real.
- For outgoing MOSIG (HKZ) traffic without silent reversal on answer, the timer-dependent simulated answering criterion is always applied.
  In digital exchange traffic with 1-bit signalling, an answering signal is always sent by the exchange.
  In digital exchange traffic with 2-bit signalling, an answering signal is seldom not sent by the exchange. A simulated answering criterion is not implemented; no call data recording takes place. Non-chargeable calls can be recognised by the register call signal B5.

10.22.1.5 Re-Seizure Prevention

For bothway exchange traffic, a re-seizure prevention timer can be set for outgoing seizures via an AMO in the HiPath 4000 system (default value 6 seconds, adjustable in steps of 0.25 seconds between 0 and 6).

10.22.1.6 Calling Line Identification Display at the Digite or Attendant Console

In direct-dialled incoming exchange traffic with MFC-R2, if the calling line identification number is transmitted by the exchange, it will be output on the Digite or attendant console display during ringing.

10.22.1.7 Call Data Recording and Evaluation

Call data records are compiled for all outgoing exchange calls, even those to non-chargeable destinations (except for calls via digital exchange lines with MFC-R2 signalling for which no answering criterion was received). The call data is evaluated according to the call duration (from answering criterion to clear back signal) and tariff rate. The tariff rate is determined by analysing the leading digit sequences of the stored destination number, even if the destination is a non-chargeable one.

In the case of calls set up by the attendant and then transferred to a HiPath 4000 user, the destination number is not stored in the call data record. These calls, which can be identified by means of the calling line identification number, are specially marked during evaluation.

10.22.2 Analog Exchange Traffic

Bothway Exchange Traffic without DID/DDI and with MOSIG (HKZ) Signalling (TMANI)

2-wire a/b exchange trunks with HKZ (MOSIG) signalling and -48V feed voltage from the public network exchange.

The trunks can be configured for bothway traffic or one-way traffic (incoming traffic only or outgoing traffic only)

No call charge pulses are transmitted by the public exchange

Secondary electrical safeguarding against voltage peaks of 1000 V, with 8 s surge and 20 s drop

Manual blocking and LED status signalling for each exchange circuit on the trunk module

Real answering criterion ("silent reversal") or simulated criterion (after timeout) for initialising call data records. Special service access code to non-chargeable bearer services with immediate switching.

a/b wire reversal possible

Dialling information signalling:pulse dialling and DTMF signalling

Ringing voltage detection:

- Frequencies:15 Hz to 30 Hz
  The ringing voltage detection at 60 $\pm$ 3 Hz is **<u>not</u>** implemented as originally intended.
- Voltage:40 $\leq V_{rms} \leq$ 90 for 25 Hz
  >34 V_{rms} for 60 Hz
- Pulse/Pause:1000/4000 ms.

Line signalling criteria for analog exchange traffic
The signalling criteria are transmitted as DC pulses via the a/b wires of the 2-wire exchange trunks

- incoming traffic
  – seizure via AC ringing voltage
  – answering (loop closure in the PABX)
  – backward release (by opening the PABX loop)
- outgoing traffic
  – seizure via loop closure in the PABX
  – exchange dialling tone as seizure acknowledgement
  – dialling information signalling: pulse dialling or DTMF signalling
  – silent reversal detection
  – forward release (by opening PABX loop)
  – backward release by a/b polarity reversal or short loop interrupt followed by polarity revesal.

The HiPath 4000 waits max. 3 minutes for exchange dialling tone.

Incoming Exchange Traffic with DID/DDI and MFC-R2 Signalling (TMLRB)

DC signalling from PABX only, not from public network exchange.

Dialling information signalling:MFC-R2

Features:

- idle line monitoring
- seizure
- dialling
- answering
- backward release

- forced backward release
- forward release
- operator re-ring by remote exchange operator
- PABX station line blocking

10.22.3 Bothway Exchange Traffic with DID/DDI (DIUN2)

Digital exchange traffic is carried out via PCM 30 highways with channel-associated signalling. Each separate channel of a PCM30 highway can be configured for incoming or outgoing traffic as desired.

Dialling information signalling:pulse dialling, DTMF and MFC-R2

Two alternate signalling types are supported:

- 1-bit pulse signalling (E&M)
  This signalling system is an adapted E&M signalling for carrier frequency paths. In contrast to the standard signalling procedure, the 'b'-bit is used for signalling.
- 2-bit signalling
  This signalling system is a channel-associated signalling system specially for PCM30 links with MFC-R2 register signalling. It is a continuous-mode signalling system based on the ITU Q.421 recommendations with additional moditications are required by Telebrás.

10.22.4 R2 Register Signalling

Only a sub-group of the possible signals of the MFC-R2 signalling system is used for exchange traffic to and from the PABX.

MFC-R2 signalling is used for DID exchange traffic and for bothway exchange traffic via digital exchange lines with E&M or 2-bit signalling systems.

The forward and backward signals and the combinations used correspond to ITU Q.440.

The meanings of the register signals differ somewhat from the ITU Q.441 recommendation and are listed below:

- Group I forward signals

I-1 . . . I-10	as per Q441 'Digit 1 . . . 0'

I-11	'Echo-cancelling on transmitting side'. Treatment by PABX: A-side: not transmitted B-side: A-4 signal used as receive acknowledgment
I-12	'Request rejected' or International transit traffic without echo cancelling on transmitting side' Treatment by PABX: A-side: transmitted as acknowledgement of A-11 ... A-15 in general, and A-1 or A7, A - 8, A-9 if requested digit not available. B-side: ignored if received as first group I signal; if acknowledgement of A-5 signal, no further A-5 signals are transmitted; if acknowledgement of A-1 signal, PABX transmits A - 4.
I-13	'Connected to testing receiver' Treatment by PABX: A-side: not transmitted B-side: A-4 signal used as receive acknowledgment
I-14	'Echo-cancelling on receiving side' or International transit traffic' Treatment by PABX: as for I - 13
I-15	'End-of-dial' if received in response to A-5 or 'Satellite connection' if received as response to AX ≠ A5. Treatment by PABX: A-side: Only sent in response to A5, if A-number transferred in full. B-side: Accepted if sent in response to A - 5; ignored if received as first group I signal or in response to AX ≠ A5.

- Group II forward signals

II-1	'standard line'
II-2	'line with call data recording'
II-3	'testing receiver'
II-4	'coin-operated telephone, national'
II-5	'long-distance operator'
II-6	'long-distance data transmission'
II-7	'coin-operated telephone, international'
II-8	'international data transmission'
II-9	'priority international caller'
II-10	'international operator'
II-11	'user with special services' is used to prevent special services from being activated for B-user, e.g. a) consultation/transfer b) conference calls c) forced call forwarding d) call forwarding on no answer e) operator intercept/attendant intercept
II-12	not assigned

II-13	not assigned
II-14	not assigned
II-15	not assigned Treatment by PABX: A-side: in response to A-5 or A-3, only II-1 is transmitted B-side: II-1 to II-11 is received, but treated as 'normal user'. Receipt of II-12 to II-15 acknowledged with A-4 or B-4.

- Group A backward signals

A-1	'Send next digit' (N+1)
A-2	a) 'echo cancelling required on destination side?' Source register responds to a) with I-14, if echo cancelling required or b) repeat first digit already transmitted' The exchange sends the signal for this digit as an alternative to signals A - 7, A-8 and A-9. Treatment by PABX: A-side: not transmitted B-side: response as described in point b)
A-3	'Transition to group B' The transmitting side responds with group II signals. The same user category is transmitted as in response to A-5. A return to group I signals is not implemented. A-3 can also be transmitted as a pulse.
A-4	'All trunks busy' Signal can be sent in response to any forward signal, if no further connection setup possible. The source system applies the busy signal to the calling line and clears down the connection. If A-4 is transmitted, it is always the last signal. A-4 can also be transmitted as a pulse.
A-5	'Transmit category and name of caller' In response to the first A-5 signal, the user category is sent back with a group II signal. In response to further A-5 signals the system switches back to group I signals in order to transmit the calling line number. When the calling line number transmission is completed, the system transmits the signal I-15. If the calling line number is not required, the system sends A-X ≠ A-5 in response to the calling line category request. If the PABX identifies the end of the calling line number, it does not wait for the I-15 signal but send A-X≠A-5 as answer to the last received digit. Treatment by the PABX: B-side: In response to a received B-number (called line), the A-5 signal is repeated until calling line and number (A-number) are complete. Exception: If I-12 or more than 10 digits are received. In both cases, the calling line number request is repeated. A-side: Response to received A-5 signals as described above.
A-6	not assigned
A-7	send second digit before last (N2)
A-8	send third digit before last (N3)

A-9	send digit before last (N1) Treatment by PABX B**-side**: A-7 to A-9 are not transmitted A-side: A-7 to A-9 are separately answered with signals I-1 to I-10. After each transmission, one or more A-1 signals will be received.
A-10	not assigned
A-11	as per ITU Q.441
A-12	as per ITU Q.441
A-13	as per ITU Q.441
A-14	as per ITU Q.441
A-15	as per ITU Q.441 Treatment by the PABX: A-11 to A-15 are not transmitted. If one of these signals is received, it is answered with I-12. If required, the signals A-3 and A-4 can be sent as pulses. A-4 can also be sent at any time, i.e. is not response-dependent. The conditions for pulse mode transmission of A-3 correspond to Figure 13 in Q.442 of the ITU Blue Book

- Group B backward signals

B-1	'Line free, call charge pulses' On receipt of answering criterion, call data recording is started
B-2	Line busy Connection cleared foward by source register
B-3	New call number Treatment by the PABX: not transmitted; instead, call is re-routed to the attendant and B-1 is transmitted
B-4	All trunks busy Connection cleared foward by source register
B-5	Line free, no call charge pulses On receipt of answering criterion, call data recording is started.
B-6	Line free with call charge pulses and caller hold option B-1 signal is sent to the local exchange instead (and thence to PABX)
B-7	Line not switched Treatment by the PABX: not transmitted; instead, call is re-routed to the attendant and B-1 is transmitted.
B-8	not assigned
B-9	not assigned
B-10	not assigned
B-11	not assigned
B-12	not assigned
B-13	not assigned
B-14	not assigned

B-15	not assigned Treatment by the PABX: B-side: Only the signals B-1, B-2 and B-4 are transmitted A-side: On receipt of B-1 or B-5, the register connection is cleared down and the voice channels of the calling line switched to the outgoing exchange trunk. All other group B signals are answered by transmitting a forward release. In some non-chargeable connections, no answering message is received in response to B-1 or B-5 when the calling line user answers.

10.22.5 Ringing Cadences

External ringing tone: 1000/4000 ms

Internal ringing tone: 350/300/350/4000 ms

Frequency: 25 ± 2.5 Hz

Ringing voltage: 70 ± 15 V_{rms}

10.22.6 Audible Tones for Brazil

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone	425	Continuous tone
Busy tone	425	**250**/250
Ringback tone	425	**1000**/4000
Internal dial tone	425	**920**/80

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse/Pause [ms]	Level (dBm0)
Internal dial tone	1	425	**920**/80	-3
External dial tone	2	425	Continuous tone	-3
Special dial tone	3	425+400	Continuous tone	-3
Ringback tone	4	425	**1000**/4000	-3
Busy tone	5	425	**250**/250	-3
Override tone	6	425	**60**/1800	-12
Call waiting tone	7	425	**200**/300/**200**/1300	-12
NU tone	8	950/1400/ 1800	Triple tone **320-320-320**/960	-3
Data call tone	9	1300	**600**/1800	-3

Tone	Tsl	Frequency [Hz]	<u>Pulse</u>/Pause [ms]	Level (dBm0)
Negative acknowledgement tone	11	425	**<u>100</u>**/100	-3
Conference auxiliary tone	24	425	Continuous tone	-12
LCR	26	1800	**<u>340</u>**/200/**<u>340</u>**/200/**<u>340</u>**/1000	- 3

Special tone sequences:

Tone	Sequence	derived from
Conference tone	200 ms 300 ms 200 ms Continuous	Conference auxiliary tone Pause Conference auxiliary tone Silence
Call waiting tone	Continuous	Call waiting tone
Call waiting tone for night service terminal	Continuous	Call waiting tone
Override tone	60 ms 100 ms Continuous	Conference auxiliary tone Pause Override tone

10.23 Greece

10.23.1 Analog Exchange Traffic

DID with F-Criteria Signalling and Pulse Dialling (TMFS)

2-wire main station interface line for voice and signalling.

Signalling method:pulse dialling

Bothway Main Station Interface Line with Pulse Dialling or DTMF (TMANI)

2-wire main station interface line for voice and signalling.

Signalling method:pulse dialling or DTMF

10.23.2 One-Way and Bothway Exchange Traffic with DID (DIUN2)

Each channel of the DIUN2 can be configured for one-way (outgoing/incoming) or bothway traffic.

1. One-way incoming exchange traffic
   - Pulse-dialling or DTMF
   - Line signalling criteria
     – Idle
     – Seizure
     – Seizure acknowledgement
     – Pulse dialling or DTMF signalling
     – End-of-dialling
     – Answering
     – Call tracing, optional
     – Forward release
     – Backward release
     – Blocking
   - Call tracing

   This feature can be activated in incoming traffic, if it is configured in the system (via AMO) and if the call is made to a HiPath 4000 user with malicious call tracing authorization (classmark).

   Call tracing starts a print-out at the service terminal with the following data:
     – Date and time
     – Call number of called party (user B)
     – PEN (hardware location address) of the voice channel of the DIUN2.

     The call tracing criterion is sent to the exchange, where the caller is then identified (A-caller)
   - If an incoming call does not provide further digits after seizing the incoming line of the PABX, the call is either
     – forwarded to the attendant (attendant intercept) or
     – released (forward release with ATB signal).
2. Outgoing exchange traffic
   - Pulse-dialling or DTMF
   - Line signalling criteria
     – Idle
     – Seizure

  - Seizure acknowledgement
  - Pulse dialling or DTMF signalling
  - Call charge pulse
  - Call tracing, optional
  - Forward release
  - Backward release
  - Blocking

- Call tracing

  When the call tracing criterion is received from the exchange, it starts a print-out at the service terminal with the following data:

  - Date and time
  - Call number of calling, internal party (user A)
  - Call number of called, external party (user B)
  - PEN (hardware location address) of the voice channel of the DIUN2.

10.23.3 Audible Tones for Greece

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone	425	**200**/300/**700**/800
Busy tone		
Ringback tone		
Internal dial tone		

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse/Pause [ms]	Level (dBm0)
Internal dial tone	1	425	**200**/300/**200**/300/**200**/800	-3
External dial tone	2	425	**200**/300/**700**/800	-3
Special dial tone	3	425+400	Continuous tone	-3
Ringback tone	4	425	**1000**/4000	-3
Busy tone	5	425	**160**/440	-3
Override tone	6	425	**200**/300/**200**/1300	-14
Call waiting tone	7	425	**100**/1900	-14

Tone	Tsl	Frequency [Hz]	Pulse/Pause [ms]	Level (dBm0)
NU tone	8	950/1400/ 1800	**320**/320/**320**/960	-3
Data call tone	9	1300	**600**/1800	1
Conference auxiliary tone	24	425	Continuous tone	-14
LCR	26	1800	340/200/340/200/340/1000	- 3

Special tone sequences:

Tone	Sequence	derived from
Conference tone	200 ms 300 ms 200 ms Continuous	Conference auxiliary tone Pause Conference auxiliary tone Silence
Call waiting tone	Continuous	Call waiting tone
Call waiting tone for night service terminal	Continuous	Call waiting tone
Override tone	Continuous	Override tone

10.24 Australia

Audible Tones for Australia

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone		
Busy tone		
Ringback tone		
Internal dial tone		

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	Pulse / Pause [ms]	Level [dBm0]
Free channel	0	0	Continuous	
Internal dial tone (=initial dial tone)	1	400 + 450	Continuous	- 11
External dial tone 1	2	425	Continuous	- 11
Special dial tone	3	400	**200**/300/**200**/300/**200**/800	- 11
Ringback tone	4	400+450	**400**/200/**400**/2000	- 11
Busy tone	5	425	**375**/375	- 11

Tone	Tsl	Frequency [Hz]	**Pulse** / Pause [ms]	Level [dBm0]
Override tone	6	425	**800**/15,000	- 11
Call waiting tone	7	425	**200**/200/**200**/4400	- 11
NU tone	8	425	**2500**/500	- 11
Data call tone	9	1300	**600**/1750	- 11
External dial tone 2	13	400 + 425	Continuous	- 11
Conference auxiliary tone	24	425	Continuous	- 11
LCR	26	1800	**340**/200/**340**/200/**340**/1000	- 4

Special tone sequences:

Tone	Sequence	derived from
Conference tone	800 ms Continuous	Conference auxiliary tone Free channel
Call waiting tone	Continuous	Call waiting tone
Call waiting tone for night service terminal	Continuous	Call waiting tone
Override tone	800 ms 500 ms Continuous	Conference auxiliary tone Free channel Override tone

10.25 Hungary

Audible Tones for Hungary

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	**Pulse** / Pause [ms]
Exchange dial tone		
Busy tone		
Ringback tone		
Internal dial tone		

The following audible tones are to be sent by the PABX:

Tone	Tsl	Frequency [Hz]	**Pulse** / Pause [ms]	Level [dBm0]
Free channel	0	0		
Internal dial tone	1			
External dial tone	2	425	Continuous	- 3
Special dial tone	3	400	Continuous	- 8

Tone	Tsl	Frequency [Hz]	Pulse / Pause [ms]	Level [dBm0]
Ringback tone	4	400+450	**1240**/3760	- 3
Busy tone	5	425	**300**/300	- 3
Override tone	6	425	**300**/300/**300**/1500	- 11
Call waiting tone = Conference tone	7	425	**40**/1960	- 11
NU tone = alerting tone (SIT)	8	950/1400/ 1800	**340**/340/**340**/1000	- 3
Waiting tone)	11	425	**120**/120/**120**/500	- 3
Positive acknowledgement tone				
Test tone	15	900	Continuous	- 4.5
Alarm tone	23	1000	Continuous	- 0.66
Conference auxiliary tone	24	425	Continuous	- 11
LCR	26	1800	**340**/200/**340**/200/**340**/1000	- 3

Special tone sequences:

Tone	Sequence	derived from
Conference tone	40 ms 40 ms Continuous	Conference auxiliary tone Pause Call waiting tone
Call waiting tone	40 ms 40 ms Continuous	Conference auxiliary tone Pause Call waiting tone
Call waiting tone for night service terminal	40 ms 40 ms Continuous	Conference auxiliary tone Pause Call waiting tone
Override tone	40 ms 40 ms Continuous	Conference auxiliary tone Pause Call waiting tone

10.26 India

Audible Tones for India

The following audible tones are to be recognised by the PABX:

Tone	Frequency [Hz]	Pulse / Pause [ms]
Exchange dial tone	400/25 (400 Hz modulated with 25 Hz)	Continuous tone
Busy tone	400	**750**/750

Tone	Frequency [Hz]	<u>Pulse</u> / Pause [ms]
ATB tone	400	**<u>250</u>**/250
Internal dial tone		

The following audible tones are to be sent by the PABX (same as UK):

Tone	Tsl	Frequency [Hz]	<u>Pulse</u>/Pause [ms]	Level (dBm0)
Internal dial tone	1	400+450	Continuous tone	-6
External dial tone	2	350+440	Continuous tone	-6
Special dial tone	3	440+480	**<u>200</u>**/300/**<u>200</u>**/300/**<u>200</u>**/800	-6
Ringback tone	4	400+450	**<u>400</u>**/200/**<u>400</u>**/2000	-11
Busy tone	5	400	**<u>380</u>**/380	-14.5
Override tone	6	1400	**<u>200</u>**/1500	-10
Call waiting tone				
NU tone	8	400	Continuous tone	-14.5
Data call tone				
Positive acknowledgement tone	10	520/400	**<u>340</u>**/**<u>340</u>**/800	-6
ATB tone	11	400	**<u>400</u>**/360/**<u>220</u>**/520	-6
Negative acknowledgement tone	14	400/520	**<u>340</u>**/**<u>340</u>**/800	-6
Conference auxiliary tone	24	1400	Continuous tone	-10
LCR	26	1800	340/200/340/200/340/1000	-11

Special tone sequences (same as UK):

Tone	Sequence	derived from
Conference tone	2000 ms Continuous	Override tone-Alerting tone Silence
Call waiting tone	100 ms 10000 ms 100 ms Continuous	Override tone-Alerting tone Pause Override tone-Alerting tone Silence
Call waiting tone for night service terminal	100 ms 100 ms 100 ms 10000 ms 100 ms 100 ms 100 ms Continuous	Override tone-Alerting tone Pause Override tone-Alerting tone Pause Override tone-Alerting tone Pause Override tone-Alerting tone Silence
Override tone	Continuous	Override tone

10.27 USA

10.27.1 National ISDN BRI

NI BRI allows the customer, as end-user, access to the HiPath 4000 system according to the protocol and procedures defined for NI BRI by Telcordia Technologies.

Each of the B-channels can be circuit-switched by the network exchange independently. Control over B-channel connections for demand applications resides in the signaling messages passed via the D-channel.

Although NI BRI provides for packet-switched data calls as well, this functionality not be supported on HiPath 4000.

For user -> network calls, the desired bearer capability and the called user's directory number be used for routing. For network -> user calls, the called user's DN and bearer capability be used for terminal selection.

If the ISDN user wishes to establish a call, the user must have equipment capable of interfacing with the 144 kbps bit stream and sending appropriate messages to request Layer 3 call setup.

General Requirements

For the functionalities the HiPath 4000 offer an ISDN Basic Rate Interface compliant with the Telcordia Technologies specifications applicable to SPCSs and consistent with Reference SR-4620: 1999 Version of National ISDN Basic Rate Interface Terminal Equipment Generic Guidelines.

NI BRI support only Class I equipment, as defined in Reference GR 268: ISDN Basic Rate Interface Call Control Switching & Signaling Generic Requirements.

Layer 1 Requirements

The 2 wire U2B1Q physical interface for NI BRI comply with Reference TR-NWT-393: Generic Requirements for ISDN Basic Access Digital Subscriber Lines, and Reference TR-NWT-397: ISDN Basic Access Transport System Requirements.

Layer 2 Requirements

Link layer D-channel call control (LAPD) comply with Reference TR-TSY-793: ISDN D-Channel Access Signaling & Switching Requirements (Layer 2).

Layer 3 Requirements
Uniform Cause Values on NI BRI

The switch can inform terminal equipment about the progress of terminal initialization procedures or of a call through D-channel signaling, using the Cause information element.

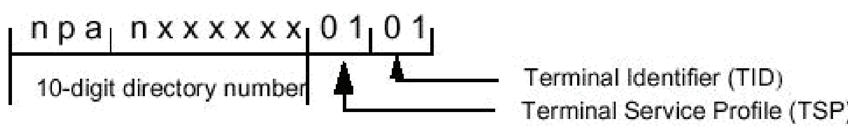
Terminal Identification and Initialization

NI BRI on HiPath 4000 support only initializing terminals.

Terminal Initialization employing Customer entered SPIDs

NI BRI terminal identification and initialization procedures which presumed manual programming of the Service Profile Identifier into the terminal equipment, whether initiated by the switch or by the terminal, comply with those defined by the requirements in Reference TR-TSY-000847: ISDN Features Common Switching & Signaling Generic Requirements.

The SPID be coded as a 9-20 digit number and not as a 3-20 digit number. Commonly, the customer uses a 10-digit DN in composing the SPID, which is programmed into the device as follows:

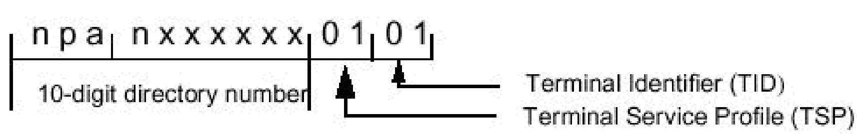


The value and format of the SPID define in section 3.6.9 of Reference TR-TSY-000847 not be constrained by HiPath 4000 design.

Terminal Initialization employing Automated SPID Selection

NI BRI terminal identification and initialization procedures using Automated SPID Selection comply with the requirements specified in Reference GR-2941: Generic Requirements for Automated SPID Selection.

The Generic SPID Format consists of 14 digits split into these components:



HiPath 4000 support as many DNs per BRI as memory has allow. During Auto-SPID Selection, however, a maximum of 8 10-digit DNs be sent to a terminal in response to the universal SPID.

The HiPath 4000 allow a device to request initialization, which includes the initiation of auto-SPID selection, at any time.

Remarks Independent of Terminal Initialization Type

As an exception to the requirements for the TSP, NI BRI employ a concept similar to that of both unique and shared TSPs.Requirements for the assignment and administration of identifiers are not supported.

Requirements for Auto-SPID Selection be interpreted on a DN basis when reference is made by Telcordia Technologies to TSPs.

Basic Call Control

Layer 3 call control comply with the requirements, GR 268: ISDN Basic Rate Interface Call Control Switching & Signaling Generic Requirements for Class I equipment for the functionality indicated in this section. The general rules for message sequencing, message processing, call reference administration and the network call states applicable to circuit-switched calls be supported.

Basic call be supported for all circuit-switched bearer capabilities; i.e.,

- Speech

- 3.1 kHz Audio

- 64 kbps unrestricted digital information rateadapted from 56 kbps

- 64 kbps unrestricted digital information

The packet mode bearer capability not be supported.

The timers applicable to basic call which be supported and be administrable Supplementary to compliance with Reference GR-268-CORE: ISDN Basic Rate Interface Call Control Switching & Signaling Generic Requirements, Layer 3 also supply the functionality provided by two features added to the U.S.Hicom for NI BRI data devices. The first of these is known as Multiple Call Reference. The second of these features is Dynamic Device Linking.

Call Origination Treatment

All call origination procedures be followed as defined in Section 5.2 of Reference /7/ GR-268-CORE: ISDN Basic Rate Interface Call Control Switching & Signaling Generic Requirements, with any qualifications or exceptions noted here.

- Support of the Operator System Access IE not be implemented.
- Support of the Transit Network Selection IE not be implemented..
- The Calling Party Number IE be subject to screening procedures, but its provision (optional or mandatory) as a subscription parameter not be supported.
- The Access Transport IEs, which are so called because they transport information end to end transparently through the network, be treated as indicated below.
- Support of the Feature Activation and Feature Indication IEs not be implemented.

The call origination procedures for all of the following be followed as per Reference /7/:

- Overlap sending mode

- Enbloc sending mode
- Interpretation of the digit address information (Keypad, Called Party Number, Transit Network Selection IEs)
- Sending a Call Proceeding indication to the calling user
- Misdialing treatments
- Originating B-channel selection, omitting any checks pertaining to subscription parameters; these not be supported.

Call Termination Treatment

Procedures for circuit-mode call delivery be followed, with these qualifications:

- Incoming call contention be supported on a physical line basis, but not be an administrable option.
- B-channel negotiation is not allowed at the terminating (basic rate) interface for Class I equipment and not be supported.
- Call offering not use the EID.
- As an exception to Reference /7/ requirement R5-82, HiPath 4000 determine if the bearer capability of the call incoming to the user is valid for the interface and not for the DN.
- Delivery of the Calling Party Number conform to ETSI Standards for CLIP and, but also be subject to requirements of the Caller ID Blocking / Unblocking, rather than to the boolean subscription parameter "Calling Party Number Delivery" indicated in Reference /7/, which not be supported.
- As an exception to Reference /7/ requirement R5-142, HiPath 4000 release call resources upon expiration of Timer T312, when T312 exceeds the expiration of the second expiration of Timer T303.

Call Clearing Requirements

General call clearing procedures for circuit-mode calls and call clearing procedures for answered circuit-mode calls be followed. As an exception to Reference /7/ requirements R5-328 and R5-329, HiPath 4000 not support Timer T402.

Error Treatment

Protocol error treatment be supported

- ·ndependent of call state
- for errors that occur during
  - call establishment at the originating or terminating interface
  - an active call

    - call clearing
- for errors not associated with a call
- for Nonlocking shift procedures

with the exception of requirements R5-393 and R5-394. No count of invalid messages received from user equipment be maintained, nor the provisioned threshold for this count be supported. Layer 3 procedures for a (Layer 2) data link malfunction also be supported.

Call Reference Administration

Call reference administration by HiPath 4000 be applied.

Channel Assignment

Channel assignment by the HiPath 4000 switch follow the procedures, with the exception that no administrable parameters be employed or supported in this regard.

Destination Address Interpretation

Destination address interpretation by the HiPath 4000 switch follow the procedures, with the exception those that apply to the Transit Network Selection and Operator System Access IEs. These IEs not be supported.

ISDN Routing

ISDN routing by HiPath 4000 follow the procedures for circuit-mode bearer capabilities, with the exception of those that apply to the Transit Network Selection and Operator System Access IEs. These IEs not be supported.

Network Determination of Interface Busy

Determination of an "interface busy" condition by the HiPath 4000 switch follow the procedures for circuit-switched calls, with the exception that no subscription parameters be consulted for this determination.

Timers

HiPath 4000 supports the administrable timer values T301, T303, T305, T306, T308, T309, T310, T312, T322, T400 and T408.

Tones and Announcements

Network provided tones be provided for speech and 3.1 kHz audio calls, but not as an administrable option.

Supplementary Service Requirements
Calling Number Identification Services

Calling Number Identification based on the ETSI Standards for CLIP and CLIR, but not in conflict with the requirements for Caller ID Blocking / Unblocking. Reference /9/, TR-TSY-860: ISDN Calling Number Identification Services may be consulted for background information on the Telcordia Technologies requirements for CNIS.

The ETSI implementation of CNIS (i.e. CLIP and CLIR) differs from the Telcordia Technologies version of the feature as follows:

- No provisioning for Calling Party Number Delivery exists.
- No discard control for the Calling Party Number exists.
- Screening of the Calling Party Number (implemented as per CLIP) uses a per BRI valid DN list.
- A default DN is supported on a per BRI basis, not on a TSP basis; TSPs, as defined exactly by Telcordia Technologies, are not supported. The default DN be based on the SPID assigned a given terminal.
- DN screening sets are not supported.
- Screening can be activated and deactivated per DN via a "special arrangement" subscription parameter.
- The "Privacy Change Allowed" subscription parameter is not supported.
- Calling Party Number presentation procedures (implemented as per CLIR) use either the Privacy Indication from the signalled Calling Party Number or support stimulus presentation restriction.
- The "Calling Party Number Type" subscription parameter is not supported.
- At most, one Calling Party Number is delivered. If screening of the number fails, the Calling Party Number is supplied from the database.

User-to-User Services

User-to-User Services based on Reference /17/, the ETSI Standard for User-to-User Signaling. Reference /11/, TR-TSY-845: User-to-User Services, may be consulted for background information on the Telcordia Technologies requirements. The ETSI implementation of User-to-User Services differs from the Telcordia Technologies version of the feature as follows:

- The "Delivery of User-to-User Service" subscription parameter is not supported.
- The " User-to-User Signaling Transfer" subscription parameter is supported.
- Trunk interworking is not supported.

- The use of PROGRESS messages for User-to-User-related activity not be supported.

Proprietary Features Supported for NI BRI Devices
Data Hunt Groups

Data hunt groups be supported for NI data devices.

A data hunt group be called by dialing a data group access number (analogous to a pilot number). A data hunt group member be called directly by its assigned directory number. If a busy data group member (or any data line) is called directly, no hunting or queuing is provided. A data line may be assigned to more than one data hunt group. Assigning data lines to data hunt groups be a system administration function. Password protection for data hunt groups not be provided, access class restrictions not be provided, nor the capability for a data group member to link (return) to or delink from (leave) its assigned hunt group by entering an access code.

Note that because the internal call queuing feature not be supported for HiPath 4000, a data line cannot be configured with permission to queue. Therefore, if all data lines in a data hunt group are busy, the calling data line (i.e., station that is associated with a data device) cannot be queued to the facility for the first available data hunt group member.

Data Access Control (Data COS)

The features described in this section are controlled by the assignment of a data COS.

Outgoing Call Queuing - Standby

When all trunks within an outgoing trunk group are busy, the calling data line (i.e., station that is associated with a data device) be queued to the trunk group until a trunk becomes available for the call, providing that the calling data line is configured to permit outgoing trunk queuing by LCR.

LCR Access

Assignment of an LCR COS for data devices causes LCR to select an outgoing trunk when the originating user dials the LCR access code.

10.27.2 Public PRI Networking

PRI is an application based on the use of T1 carrier as a transport medium, and as such supports standard 24 channel configurations and framing formats on 1.544 Mbps facilities. PRI is an ISDN usernetwork interface structure composed of multiple bearer channels (B-channels) and one signaling channel (D-channel). When a PRI using Facility Associated Signaling (FAS) is provided, the channel configuration is 23 B + D.

Applicable PRI Switch Vendor Specifications (Normative References)

Listed below are the current versions of the U.S. public network switch vendor specifications that are required for the development of U.S. PRI on the HiPath 4000.

a) AT&T 4ESS

1. TR41459, 'AT&T Network ISDN PRI and Special Application Specification, User-Network Interface Description', June 1999 /11/
2. TR50075, 'AT&T Toll Free Transfer Connect (SM) Service', May 1995 /12/
3. TR54016, 'Requirements for Interfacing Digital Terminal Equipment to Services Employing the Extended Superframe Format', September 1989 /13/
4. TR62411, 'ACCUNET(r) T1.5 Service Description and Interface Specification', December 1990 /14/.

b) MCI DMS-250 / DEX600

1. 014-0018-04.3F-ER, 'Network Interface Requirements for MCI ISDN PRI', Revision 4.3F, September 18, 1998 /15/
2. 014-0034-01.4F-ER, 'ISDN-based Network Call Transfer, CPE Call Processing Requirements', Revision 1.4, April 17, 1996 /16/
3. 014-0036-01.3D-ER, 'ISDN-based User-to-User Signaling Services, CPE Call Processing Requirements', Revision 1.3, February 10, 1997 /17/

c) SPRINT DMS-250 (using AT&T 4ESS protocol emulation at Layer 3)

1. Nortel NIS-A211-1, "DMS-100 ISDN PRI User-Network Interface Specification", Version NA011 Standard 08.01, August 1998 (Note: Chapters 2 and 3 of NIS-A211-1, which define Nortel's Layer 1 and Layer 2 requirements for PRI, are also applicable to SPRINT DMS-250 PRI.) /19/
2. Sprint Document (no control number), "DMS-250 PRA Interworking with 1988 41449", (defines Sprint's DMS-250 PRI option that provides "AT&T 4ESS PRI protocol emulation" at Layer 3) /18/

d) Telcordia Technologies (formerly Bellcore) NI-2 PRI

1. TR-TSY-000754, 'ISDN Primary Rate Access Transport System Requirements (Layer 1 Requirements) ', Issue 1, July 1990 /20/
2. TR-TSY-000793, 'ISDN D-Channel Exchange Access Signaling and Switching Requirements (Layer 2 Requirements), Issue 1, October 1988, plus Revision 1 (October 1994), and Bulletin 1 (November 1994) /21/
3. SR-4619, '1999 Version of National ISDN PRI CPE Generic Guidelines', Issue 1, December 1998 /22/
4. GR-2856-CORE, 'Generic Requirements for D-Channel Message Performance Monitoring and Control of ISDN Interfaces', Issue 1, July 1994 /23/

e) ITU-T Recommendations:

HiPath 4000comply with the ITU-T recommendations for PRI D-channel layer 3 signaling protocol across an ISDN usernetwork interface as described in ITU-T Recommendations Q.931 (I.451) and Q.932 (I.452) to the same extent that these recommendations have been deployed by U.S. Inter-Exchange and Local Exchange Carriers.

f) ANSI Standards:

HiPath 4000comply with the American National Standards for PRI D-channel layer 3 signaling protocol across an ISDN usernetwork interface as described in ANSI T1.607 and ANSI T1.610 to the same extent that these standards have been deployed by U.S. Inter-Exchange and Local Exchange Carriers.

PRI Layer 1

PRI Transmission Requirements

Speech and 3.1 kHz audio information transmitted to/from the U.S. Public Switched Telephone Network (PSTN) via a HiPath 4000PRI B-channel be encoded in accordance with the mylaw companding algorithm and the loss level plan requirements.

Call progress tones transmitted to the U.S. PSTN prior to the return of answer supervision comply with the limitations defined in FCC Part 68. In particular, only the following inband tones and announcements are allowed by FCC Part 68 rules and regulations to be sent to a PRI B-channel of an incoming call prior to returning answer supervision:

– Audible Ring Tone
– Busy Tone
– Reorder Tone (also known as "fast" busy tone)
– DID intercept recorded announcement, but only if it is a standard generic intercept announcement message that cannot be modified or customized by the customer.

PRI Idle Codes

PRI idle codes to be sent to the U.S. PSTN by the HiPath 4000are summarized in:

PRI Protocol Variant	B-channel (Note 1)	D-Channel	EOC (Note 2)
AT&T (4ESS)	At least seven contiguous ONEs (7F, FE or FF)	HDLC flag (7E)	FF (Note 3)
MCI (DMS-250/DEX600) Sprint (DMS-250)	7F 7F	HDLC flag (7E) HDLC flag (7E)	HDLC flag (7E) HDLC flag (7E)
NI-2 PRI	At least three ONEs in every timeslot	HDLC flag (7E)	HDLC flag (7E)

NOTES:

1. B-channel idle code requirements can be satisfied by sending 7F for all PRI protocol variants.

2. EOC idle code is dependent on the active protocol variant (in Digital Trunk Configuration)

3. AT&T TR41459 /11/ requires the EOC idle code to be "all ones" (FF) if the enhanced facility maintenance functions as defined in AT&T TR54016 /23/ are not supported, and "HDLC flag" (7E) if they are supported. Since HiPath 4000does not support these functions, FF must be sent.

PRI D-Channel Layer 2

Four Layer 2 variants are required to support the U.S. PRI variants in Hicom Unity; 1) ITU, 2) AT&T, 3) Nortel, and 4) Bellcore. The requirements for these four Layer 2 variants are specified in the following documents:

1. Layer 2 Variant = ITU-T Recommendation Q.921

2. Layer 2 Variant = AT&T TR41459 /11/, 'AT&T ISDN PRI Specification' (Part II)

3. Layer 2 Variant = Nortel NIS-A211-1 /19/, 'ISDN PRI User-Network Interface' (Chapter 3)

4. Layer 2 Variant = Bellcore:

   Telcordia Technologies (formerly Bellcore) SR-NWT-004619 /22/, 'ISDN Primary Rate Interface Generic Guidelines for Customer Premises Equipment' (Section 4), and

   Telcordia Technologies (formerly Bellcore) TR-TSY-000793 /21/, 'ISDN D-Channel Exchange Access Signaling and Switching Requirements (Layer 2)'

The active Layer 2 variant be determined based on the assigned Layer 3 protocol.

PRI Calling Services

HiPath 4000provides the following PRI access to the IEC and LEC public network calling services identified in the following.

	IEC (see Note 5)				LEC
	AT&T	MCI (Note 1)		SPRINT DMS-250	NI-2
	4ESS	DMS-250	DEX-600	(Note 2)	PRI
Incoming-only Calling Services					
In-WATS (e.g., '800/888/877' calls) - Toll-Free MEGACOM / Multimedia (AT&T) - MCI 800/888/877 (MCI) - Ultra 800/888/877 (Sprint) - In-WATS (LEC)	X	X	X	X	X
Pay-per-call service (e.g., '900' calls) - AT&T MultiQuest (AT&T) - MCI 900 (MCI)	X	X	X	-	-
Int'l In-WATS (e.g., '800/888/877' calls) - International Toll-Free (AT&T) - MCI 800 (MCI)	X	X (3)	X (3)	-	-
Outgoing-only Calling Services					
Out-WATS - MEGACOM (AT&T) - PRISM (MCI) - Ultra WATS (Sprint) - Out-WATS (Unbanded) (LEC)	X	X	X	X	X
Multiband Out-WATS	-	-	-	-	X
Access to LEC Operator	X (4)	-	-	-	X
Access to default IEC Operator	X	X	X	X	X
Equal Access to another IEC Operator	- (4)	-	-	-	X
Equal Access to IEC Basic Calling Service (Transit Network Selection / 4-digit CIC)	X (4)	-	-	-	X
ISDN Access to IEC Specialized Calling Services (via LEC PRI) (i.e., Network-Specific Facilities selection to a designated IEC)	-	-	-	-	#
Access to Hotel/Motel service operator service system	-	-	-	-	#
Selective Class of Service Screening (SCOSC) by an operator service system	-	-	-	-	#

LEGEND for PRI calling services:

X = Supported by PRI protocol; supported by HiPath 4000
# = Supported by PRI protocol; not supported by HiPath 4000
- = Not supported by (or not applicable to) the PRI protocol
(n) = See note n

NOTES FOR PRI calling services:

1. MCI PRI be supported as two PRI protocol variants in order to accommodate the PRI protocol differences between the DMS-250 and DEX600 switching platforms.

2. HiPath 4000has support SPRINT DMS-250 PRI using Sprint's Layer 3 protocol variant which emulates the AT&T 4ESS PRI protocol.

3. International In-WATS service is supported by MCI as part of their MCI 800 service. It is not treated as a separate service as it is with AT&T.

4. A potential new AT&T service, "AT&T Digital Link", would allow AT&T 4ESS PRI customers to make local calls via AT&T 4ESS PRI, whereby AT&T would operate as an LEC. This service is planned to be a new AT&T provisioning option for use in conjunction with SDN and/or Toll-Free MEGACOM calling services. As of June 1999, AT&T's TR41459 /11/ adds support for the Transit Network Selection (TNS) IE to meet the legal requirement to allow a customer to choose another IEC for a call, i.e., "Equal Access". The TNS IE takes

precedence over the NSF IE if both are present in a SETUP message and the CICs conflict. If there is no TNS IE and no CIC indicated in the NSF IE, the call is route to the default IEC.

5. HiPath 4000supports only "senderized" operation for tie trunk access via NI-2 PRI. Senderized operation is where all of the called party number digits are sent/received 'en bloc' in a D-channel SETUP message. "Cutthrough" operation, whereby called party number digits are sent/received via DTMF in the selected B-channel, is not supported for tie trunk access via PRI.

6. These PRI protocols support switched digital services in combination with other calling services (e.g., VPN) and not as a separate calling service. NI-2 PRI does not support the 64r (64 kbps restricted digital information) service.

AT&T 4ESS PRI

AT&T 4ESS PRI is a userside trunk access interface that conform to L1, L2 and L3 requirements as specified by AT&T.

- AT&T 4ESS PRI calling services to be supported by HiPath 4000:
- Toll-Free MEGACOM / Multimedia (incoming '800/877/888' domestic voice/data service)
- AT&T MultiQuest (incoming '900' pay-per-call service)
- International Toll-Free Service (incoming '800/877/888' international voice service, e.g., from Canada)
- Long Distance Service (LDS) / TSAA / AVA (incoming LEC bypass service)
- MEGACOM (outgoing voice service)
- Software Defined Network (SDN) / Software Defined Data Network (SDDN) / Global SDN (GSDN) (bothway VPN service)
- ACCUNET Switched Digital Service:
  - ACCUNET SDS 56 (bothway 56 kb/s data rate-adapted to 64 kb/s service)
  - ACCUNET SDS 64r (bothway 64 kb/s restricted data service)
  - ACCUNET SDS 64c (bothway 64 kb/s clear channel / unrestricted data service)
- Access to AT&T operator

AT&T 4ESS PRI supplementary services to be supported by HiPath 4000:

- Call-by-Call Service Selection
- Dedicated Service Selection
- Dialed Number ID Service (DNIS)

- Calling Party Number (CPN) to network
- Calling Party Number (CPN) / Billing Number (BN) from network:
  - Provisioned delivery of CPN/BN on every call
  - Requested delivery of CPN/BN on a per call basis
- Connected Number to/from network
- Redirecting Number to/from network
- NOTE: Redirecting Number to/from network as per current Hicom 300E IM implementation has be supported. Redirecting Number per IBM's message center application as supported in Hicom 300E US has not be supported in HiPath 4000
- - Alternate Destination Call Redirection (ADCR)
- - AT&T Transfer Connect (DTMF inband version only)
- - User-to-User Signaling to/from network
- NOTE: Only Message Associated User-User Information (MA-UUI) for the transport of ACL "User Data" to/from the network has be supported. Call Associated Temporary Signaling Connections (CA-TSC) and Non-Call Associated Temporary Signaling Connections (NCA-TSC), which are used in Hicom 300E US for the CorNet-VN application, has not be supported in HiPath 4000.
- - B-channel Management (SERVICE / SERVICE ACK messages)

MCI DMS-250 PRI

MCI DMS-250 PRI is a userside trunk access interface that conform to L1, L2 and L3 requirements as specified by MCI for the DMS-250 switching system.

- MCI DMS-250 PRI calling services to be supported by HiPath 4000:
- Basic access (bothway POTS service)
- MCI 800/888/877 (incoming '800/888/877' voice/data service)
- MCI 900 (incoming '900' pay-per-call service)
- PRISM (outgoing voice/data service)
- Vnet/Vision (bothway VPN service)
- Access to MCI operator

MCI DMS-250 PRI supplementary services to be supported by HiPath 4000:

- Call-by-Call Service Selection
- Dedicated Service Selection

- Dialed Number ID Service (DNIS)
- Calling Party Number to/from network
- Connected Number to/from network
- Redirecting Number to/from network NOTE: Redirecting Number to/from network as per current Hicom 300E IM implementation has be supported. Please note that DMS-250 supports sending/receiving Redirecting Number IE in only the ALERTING message.
- Network Call Redirection (NCR)
- Network Call Transfer (NCT)
- User-to-User Signaling NOTE: Only Message Associated User-User Information (MA-UUI) for the transport of ACL "User Data" to/from the network has be supported. Call Associated Temporary Signaling Connections (CA-TSC) and Non-Call Associated Temporary Signaling Connections (NCA-TSC), which are used in Hicom 300E US for the CorNet-VN application, has not be supported in HiPath 4000.
- - B-channel Management (SERVICE / SERVICE ACK messages)

MCI DEX600 PRI

MCI DEX600 PRI is a user-side trunk access interface that conform to L1, L2 and L3 requirements as specified by MCI for the DEX600 switching system.

MCI DEX600 PRI calling services to be supported by HiPath 4000:

- Basic access (bothway POTS services)
- MCI 800/888/877 (incoming voice/data service)
- MCI 900 (incoming '900' pay-per-call service)
- PRISM (outgoing voice/data service)
- Vnet/Vision (bothway VPN service)
- Access to MCI operator

MCI DEX 600 PRI supplementary services to be supported by HiPath 4000:

- Call-by-Call Service Selection
- Dedicated Service Selection
- Dialed Number ID Service (DNIS)
- Calling Party Number to/from network
- Connected Number to/from network
- Network Call Redirection (NCR)

– Network Call Transfer (NCT)

– User-to-User Signaling NOTE: Only Message Associated User-User Information (MA-UUI) for the transport of ACL "User Data" to/from the network has be supported. Call Associated Temporary Signaling Connections (CA-TSC) and Non-Call Associated Temporary Signaling Connections (NCA-TSC), which are used in Hicom 300E US for the CorNet-VN application, has not be supported in HiPath 4000.

– B-channel Management (SERVICE / SERVICE ACK messages)

SPRINT DMS-250 PRI (using AT&T 4ESS emulation at Layer 3)

SPRINT DMS-250 PRI (using AT&T 4ESS emulation at Layer 3) is a user-side trunk access interface that conform to L1 and L2 requirements as specified by Nortel Networks for the DMS-250 switching system, and L3 requirements as specified by AT&T for the 4ESS switching system for the subset of calling services and supplementary services supported by SPRINT.

-SPRINT DMS-250 PRI calling services to be supported by HiPath 4000:

– Ultra 800/888/877 (incoming '800/888/877' voice/data service)

– Ultra WATS (outgoing voice/data service)

– VPN (bothway VPN service)

– Access to SPRINT operator

SPRINT DMS-250 PRI supplementary services to be supported by HiPath 4000:

– Call-by-Call Service Selection

– Dedicated Service Selection

– Dialed Number ID Service (DNIS)

– Calling Party Number to/from network

– Connected Number to/from network

– Redirecting Number to/from network: NOTE: Redirecting Number to/from network as per current Hicom 300E IM implementation has be supported.

– B-channel Management (SERVICE / SERVICE ACK messages)

NI-2 (National ISDN - 2) PRI

NOTE: The terminology "NI-2 PRI", refers to the National ISDN PRI protocol defined by Telcordia Technologies (formerly Bellcore). "NI-2" not be construed to imply a particular version of National ISDN. The version of NI PRI which is required for HiPath 4000 is known as the "1999 Version of National ISDN PRI" by Telcordia Technologies.

The NI-2 (National ISDN - 2) PRI user-side trunk access interface conform to L1, L2 and L3 requirements as specified by Telcordia Technologies for Customer Premises Equipment. This trunk type facilitates PRI connectivity to the public network via a LEC's NI-2 PRI compliant public network switch (e.g., 5ESS, DMS-100, EWSD, or GTD-5 switching system), allowing HiPath 4000customers to access that LEC's public network calling services and supplementary services.

NI-2 PRI Calling services to be supported by HiPath 4000:

- - LEC basic access (bothway POTS service)
- - In-WATS (incoming toll-free voice/data service)
- - Out-WATS (outgoing voice/data service)
- - Multiband Out-WATS (outgoing voice/data service)
- - Foreign Exchange (FX) (bothway voice service)
- - Tie Trunk Access (TTA) (bothway voice service)
- Operator System Access
  - Access to LEC operator
  - Access to default IEC operator
- Equal Access to other IEC operator: NOTE: NI-2 PRI operator system access requires support of U.S. codeset 5 (national-specific) Operator System Access (OSA) information element
- 4-digit CIC 'Equal Access' to IEC long distance calling service (Transit Network Selection)

NI-2 PRI supplementary services to be supported by HiPath 4000:

- Call-by-Call Service Selection
- Dedicated Service Selection
- Dialed Number ID Service (DNIS)
- Calling Party Number to/from network
- Redirecting Number to/from network
- B-channel Management (SERVICE / SERVICE ACK messages)

Features:

Public Network-based Supplementary Services

HiPath 4000 provide PRI access to the IEC and LEC public network-based supplementary services identified in Table 3.

	IEC				LEC
	AT&T	MCI (Note 1)		SPRINT DMS-250 (Note 2)	NI2
	4ESS	DMS-250	DEX-600		PRI
Call-by-Call Service Selection	X	X	X	X	X
Dedicated (Pre-Provisioned) Service Selection	X	X	X	X	X
Equal Access (CIC) / Transit Network Selection	X	-	-	-	X
Operator System Access (using functional signaling)	X	-	-	-	X
DNIS (Dialed Number ID Service)	X	X	X	X	X
CLID (CPN) to Public Network (on outgoing calls)	X	X	X	X	X
CLID (CPN/BN) from Public Network (on incoming calls): - Provisioned delivery of CPN/BN on every call - Requested delivery of CPN/BN on a per call basis	 X X	 X -	 X -	 X -	 X -
CLID delivery from Public Network on All B-channel Busy (ABB) condition	-	#	#	(3)	-
Calling Party Subaddress	#	#	#	#	#
Called Party Subaddress	#	-	-	-	#
Connected Line ID (COLI)	X	X	X	X	-
Redirecting Number (number transport & delivery)	X(4)	X(4)	X(4)	X(4)	X(4)
Network Name Displays	-	-	-	(3)	#
Network Message Waiting (service/notification)	-	-	-	-	#
Network Ring Again (Callback)	-	-	-	(3)	-
Network ACD	-	-	-	(3)	-
1+ and 0+ Call Redirection	#	-	-	-	-
Information Indicator (II) Digits Del.	#	-	-	(3)	-
Information Forwarding 3 (INFO3)	#	-	-	-	-
Caller Information Forwarding (CINFO)	#	-	-	-	-
Look-Ahead Interflow	#	-	-	-	-
Call Redirection: - Alternate Destination Call Redirection(AT&T) - Network Call Redirection (MCI) - Network Redirection and Reason (SPRINT)	X	X	X	(3)	-
Call Deflection	-	-	-	-	# (NI-2000)
Business Group Networking	-	-	-	-	#
Flexible Billing Feature	#	-	-	-	-
Credit Checking Application Feature	#	-	-	-	-
Call Transfer: - 800 Transfer Connect (AT&T) - Network Call Transfer (MCI) - Two B-Channel Transfer (NI-2 PRI)	X(5)	X(6)	X(6)	-	#
User-to-User Signaling - MA-UUI - CA-TSC - NCA-TSC	 X(7) # #	 X(7) # #	 X(7) # #	 - - -	 - - -
B-Channel Negotiation	X	X	X	-	X
B-Channel Management (SERVICE Messages)	X	X	X	X	X
PRI Span Initialization Restart Procedures	X	X	X	X	X
Non-Facility Associated Signaling (NFAS)	#	#	#	#	#
D-Channel Backup	#	#	#	#	#

Table 3: Public Network-based PRI Supplementary Services

LEGEND FOR TABLE 3:

X = Supported by PRI protocol; supported by Hicom Unity
# = Supported by PRI protocol; not supported by Hicom Unity
- = Not supported by (or not applicable to) the PRI protocol
(n) = See note n

NOTES FOR TABLE 3:

1. MCI PRI be supported as two PRI protocol variants in order to accommodate the PRI protocol differences between the DMS-250 and DEX600 switching platforms.

2. HiPath 4000support SPRINT DMS-250 PRI using Sprint's Layer 3 protocol variant which emulates the AT&T 4ESS PRI protocol.

3. These supplementary services are supported by Sprint only on their 'native'

Layer 3 PRI protocol (i.e., Nortel PRI protocol). These supplementary services are not offered by Sprint when using their Layer 3 protocol option which emulates the AT&T 4ESS PRI protocol, and therefore are not supported by Hicom Unity.

4. Redirecting Number to/from network as per current Hicom 300E IM implementation has be supported. Redirecting Number per IBM's message center application as supported in Hicom 300E US has not be supported in HiPath 4000.

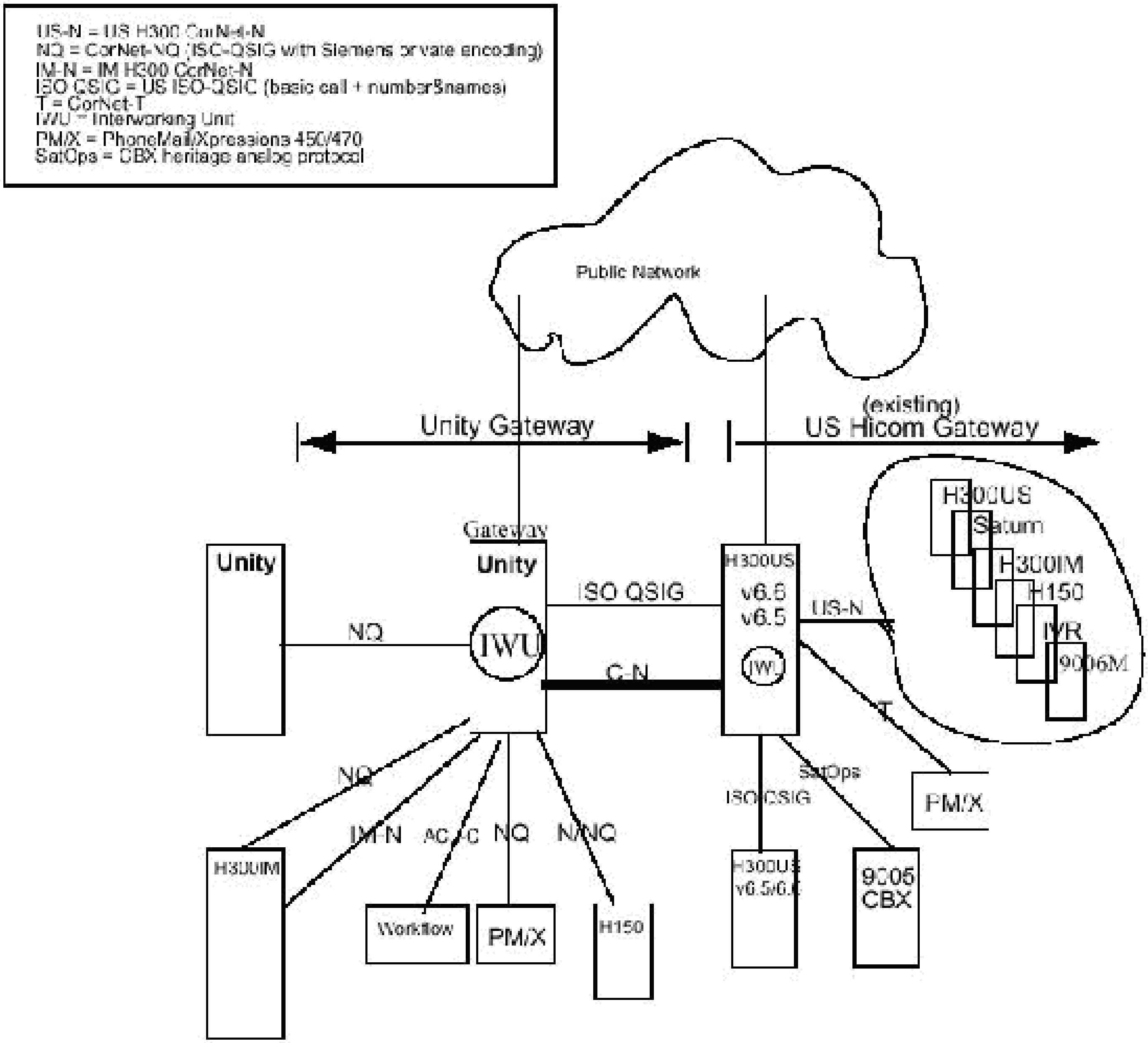
5. AT&T's 800 Transfer Connect feature (non-ISDN version) is invoked during an active two-party call using an in-band DTMF access code (*T) sent by the user to the network. HiPath 4000 has not support the ISDN signaling method of invocation, nor the "Data Forwarding" aspect of this feature.

6. Only the "basic" version of MCI's Network Call Transfer is supported by HiPath 4000. MCI's Enhanced Network Call Transfer, which allows for the transfer of two calls that are not on the same MCI PRI span and supports notification of transferred call clearing.

7. Message Associated User-to-User Information (MA-UUI) is supported in conjunction with the CallBridge for Workgroups application for the purpose of passing "User Data" between CallBridge Hosts across AT&T SDN and MCI Vnet/Vision connections..

10.27.3 Private ISDN Networking

HiPath 4000 provide gateway interworking to US Hicom and to other supported protocols.



HiPath 4000 Private Network Connectivity

CorNet-NQ be the preferred layer 3 protocol for Siemens private networks (i.e., HiPath 4000 to HiPath 4000 use the CorNet-NQ protocol, not CorNet-N). This variant provides all Siemens QSIG enhanced services based on HiPath 4000 base code plus new QSIG services (e.g., for support of PM/Xpressions). This variant also provides interworking/backward compatibility with a v6.5/6.6

– US Hicom ISO QSIG and

– CorNet-N (i.e., US Hicom)

– IM CorNet-NQ (i.e., International Market Hicom)

Description of interworking functionality:
ACD-IVR

HiPath 4000 may be a transit between two or more US Hicom's with IVR connectivity. HiPath 4000 transparently transmit the ACD-IVR either over CorNet-N or tunneled over CorNet-NQ

Call Completion No Reply/Station Message Waiting Callback

1. HiPath 4000 has this one key and an LED which indicates that a message reminder (or PhoneMail message) has been left. The US Hicom has separate callback keys for MWI and CCNR. HiPath 4000, as the terminating PINX, makes a decision (only HiPath 4000-HiPath 4000 over CorNet-N) on whether CCNR or MWI applies based on the called party's device MW capabilities (i.e., MWCB indicator).

2. US Hicom is originator: When HiPath 4000 receives a MWI message from a US Hicom, HiPath 4000 activate MW callback/reminder for user B and turn the LED on. When HiPath 4000 receives a CCNR invocation, HiPath 4000 activate CCNR for user B but not turn the LED on.

3. HiPath 4000 is originator: When the HiPath 4000 user reaches a ringing user on a US Hicom and presses the callback button, if the US Hicom user has a mailbox , a MWI invocation will be sent. Otherwise, if the US Hicom user does not have a mailbox, a CCNR invocation will be sent by HiPath 4000.

4. Cancellation of MWI (not CCNR) from the terminating user to the originating user's PINX is supported.

5. CCNR Mailbox interworking/No CCNR Mailbox interworking is sent in the backward direction from HiPath 4000 to indicate whether "Briefkastenruckruf" is implemented at the terminating PINX.

6. The MW/CCBS/NR message queue is not saved over restart.

Call Forwarding/Diversion

1. HiPath 4000 support two network CF variants for CF internal/external and CF DND internal and external.

2. Delayed CF on busy, be implemented locally. The CF busy execution is delayed based on timer expiry, therefore, the CF signalling in CorNet-NQ follow the CF delay signalling (not busy signalling) although the diversion reason be CF on busy (cfb).

Call Park / Call Pickup

1. Network Call Park Interworking between HiPath 4000 NQ Call Park to US Hicom CorNet-N is realized.

2. For park retrieval on recall, ACL-C is required to distinguish in the DISConnect message between a

· call abandon (normal call hang-up)

· park recall on no answer and

· park recall on busy.

Classmarks and Common Information (CMN)

1. CMN be supported in CorNet-NQ per ECMA standards.

2. Private CMN encoding/extension not be provided, Classmarks be extended as follows instead.

a) FeatureID

· NoTonesForUnsuccessfulCall

· Disconnect and answer supervision

· SilentMonitoringAllowed

· RccbsAllowed = yes

· can/cannot off-hook trunk queue

· can/cannot be conference member

b) CallHistory be used and extended as part of Call Transactions

Message Waiting

1. Message Waiting Indication (MWI) is implemented according to ISO QSIG for voice mail/message center servers.

2. MWI service is also used for non-call related station message waiting.

3. HiPath 4000 provide interworking between CorNet-N and ISO QSIG MWI for the MWI service for interworking with US Hicoms.

Near End Tone Generation

1. HiPath 4000 injects busy tone into the B-channel when it's user is busy/ OOS and sends inband tones/announcements in the Progress message, a Classmark codepoint is sent. This codepoint indicates whether the terminating PINX responds to a busy/OOS condition with no tone injected into the B-channel or that the terminating PINX supplying tone from the far end. The receiving HiPath 4000 honor the Classmark.

2. For outgoing calls, on a trunkgroup basis, a parameter indicates in which way to the Classmark codepoint is set. For US Hicom CorNet-N trunks, the default value is set to no tone.

3. For incoming calls, when this codepoint is not received (e.g., from a US Hicom), the trunk parameter be applied (i.e., no tone for US Hicom inc. trunks). If the call is tandeming through a HiPath 4000 gateway it add the trunk parameter value in the Classmarks before transmittal.

Private Cause Values

Private cause values, if applicable, be tunnelled and the codeset 0 Cause IE ITUT cause value = normal, unspecified. The receiving PINX process the private cause value appropriately when received.

Remote Call Completion on Busy Subscriber (R-CCBS)

1. A PhoneMail/Xpressions calling user is the only known invokee of this feature.

2. CorNet-N R-CCBS equivalency be implemented in NQ and interworked with CorNet-N. The R-CCNR procedure is not implemented.

3. Backward compatibility is realized with CorNet-N US Hicom R-CCBS.

4. HiPath 4000 CorNet-NQ baseload currently supports connection retention and connection release required for this R-CCBS service.

5. A Cornet Classmark for can/cannot perform remote CCBS currently exists. This is an attribute of the switch and not the user.

Simple Dialog: Keypad and Display Information

1. HiPath 4000 support the requirements of the Client User and transit PINX while servers (e.g., Xpressions 450) support the requirements of the Served.

2. Private extensions are not been identified.

3. It is optional for the Client User PINX and Served user PINX to act as an incoming or outgoing gateway PINX for interworking with a public ISDN for SS-SD.

4. Support of relevant ECMA-143 and ECMA 165 at the Client User PINX and Served user PINX is realized.

Single Step Call Transfer (SS-SSCT)

1. SS-SSCT based on currend ECMA standard is implemented to support PhoneMail/Xpressions connections.

2. Postdialing and digitinfo procedures are not implemented.

3. Interworking with CorNet-N Remote Call Transfer is supported.

10.27.4 Analog Trunking

The term "US Hicom" in this part is meant to include all 9006i/US Hicom 300E releases V6.1 through V6.6 unless otherwise specified with a version level. T1 Digital (non-ISDN) and Analog Trunk Interfaces - CO are conform to /4/ ANSI Standard EIA/TIA-464B, 'PBX Switching Equipment for Voiceband Application'.

– Conventional analog tie be supported for non-ISDN private networking.

– CorNet NQ or CorNet-N are the preferred private networking products.

- SatOps not be supported on HiPath 4000. US Hicom provides interworking between CorNet-N and SatOps, therefore, heritage CBXs may still be supported in a HiPath 4000 network when the US Hicom interworks with HiPath 4000 via CorNet-N.
- Local station features (e.g., hold, transfer, conference) on the HiPath 4000 interact with users over analog trunks similar to the feature interaction on US Hicom.
- The following types of analog and T1 (robbed bit signalling) emulation trunk types be supported: E & M, Loop Start (LS), and Ground Start (GS).
- Both Dial Pulse (DP) and Dual Tone Multi-Frequency (DTMF) addressing methods be supported.
- T1 digital (non-ISDN) interfaces conform to the electrical and physical characteristics of the DSX1 (Digial Cross-Connect type 1) type interface.
- HiPath 4000 support the following types of trunks for access to/from US public telephone networks:
  - Ground Start Trunks (including ground start FX/WATS trunks)
  - Loop Start Trunks (including loop start FX/WATS trunks)
  - Direct Inward Dialing (DID) Trunks
- One-way 911 CAMA Access Interface not be implemented.
- Prefix digits may be applied to any public analog/T1 incoming CO trunk (e.g. DID).

Trunk to Trunk (TTT) Connections/Transfer

1. User and trunk permissions be provided and no trunk be hung beyond a user-administrable time limit.

2. Two trunks involved in the trunk-to-trunk connection may be directly connected to the node of the transferring station or, one or both trunks may be indirectly connected to the transferring node via a conventional tie trunks (tandem/transit connection) or CorNet trunks.

3. One of the two trunks must be answered (declared answered) before the station or attendant can initiate the transfer to the other trunk. Declared answer (and answer supervision) occurs when the trunk does not have true answer supervision. In this case, answer is declared (i.e., simulated) and a recall timer is started to prevent the trunk from being hung if not answered within the timer limit.

4. Transfer after answer of both connections may also occur and disconnect supervision is realized when both trunks do not provide reliable disconnect.

5. A station user can establish such a connection even if proper (reliable) disconnect supervision of the resulting connection cannot be assured. If reliable disconnect supervision cannot be assured, then the connection may

be allowed (if trunks and user classmarks permit), with the use of the automatic trunk-to-trunk recall timer. Upon expiration of the recall timer (when used), the call is recalled to an attendant console so that an attendant can manually supervise the connection

6. Interworking with CorNet-N disconnect and answer supervision is implemented.

7. A properly classmarked station or attendant can establish a properly classmarked trunk-to-trunk connection between an

· incoming trunk call and an outgoing trunk,

· incoming trunk call and incoming trunk, and

· outgoing trunk call and outgoing trunk.

8. Answer supervision applies to non-ISDN outgoing CO trunks only. Disconnect supervision applies to non-ISDN incoming and outgoing CO trunks. In the US, E&M tie trunks always provide disconnect supervision but not answer supervision.

Feature Interaction

– A station can enter into consultation mode or conference state before performing a trunk to trunk transfer.

– When a station is not authorized to make a trunk to trunk connection, the held trunk recalls the transferring station or an attendant if the station is not available.

– Immediate recall occur if the transfer was unsuccessful.

– If true answer supervision is provided by the destination outgoing trunk, the transferred trunk recalls the transferring party upon expiration of the noanswer timer.

– Trunk to Trunk connections has recall the Attendant dependent on trunk disconnect supervision.

– Trunk-to-trunk connections NOT be allowed for data calls.

DIT Dedicated Incoming Trunks

A dedicated incoming trunk call applies to CO and tie trunks and be routed directly to a predefined ringdown station. DIT can coexist with other types of trunks in the system (e.g., DID).

1. When seized, an incoming trunk marked as a DIT has bypass the normal attendant handling and terminate to a preassigned private network user. No called number is delivered.

2. Calls to a busy station be able to be automatically camped-on, with standard camp-on recall rules.

3. DITs can be used for outgoing calls, using the standard trunk access methods, class of service restrictions, code restriction and trunk group assignments.

4. A station be assignable to more than one DIT.

5. The DIT destinations may reside in the entry point HiPath 4000 or elsewhere in the network. Destination station numbers may be an index to the hotline destinations which are shared between DIT, off-hook alarm and hotlines.

6. DITs may be in a common outgoing group but in discrete hunt groups at the CO end. Regular incoming trunks are not be allowed to hunt to a DIT group at the CO end. Arrangements are made with the common carrier to accomplish this result.

7. Answer supervision not be returned to the CO until the call is answered . Connections to voice recorded announcements return answer supervision if the call has not yet been answered. Connections to outgoing CO trunks return answer supervision when outpulsing to the trunk has been completed.

8. If the assigned DIT station is out of service or if the assigned station number is vacant, the incoming DIT call be routed to the attendant or to the night answer arrangement if no attendant is in service.

9. If the DIT station is classmarked for originate only service, the incoming DIT call be routed to the attendant or to the night answer arrangement if no attendant is in service.

10. If the DIT station is classmarked for DID blocking, the incoming DIT call be allowed to terminate.

11. If the assigned station is not authorized to receive the call due to ITR group restriction, the incoming DIT call be routed to the attendant or to the night answer arrangement if no attendant is in service.

12. Answer supervision be returned to a DIT trunk when the destination party has answered the call or when the call is connected to a recorded announcement and the call has not yet been answered.

Feature Interaction

– All call forwarding functions apply to DITs, including the transferring of DIT call to the attendant when the console is in service.

– The assigned DIT station may be a master of a station hunt group.

– The number assigned to the DIT trunk may be a pilot hunt group number.

– The assigned station may be a member of a pickup group.

– If the assigned station transfers the incoming call and hangs up, and the third party does not answer the call within the noanswer recall time, the call be handled in accordance with the standard procedure for station transfer of trunk calls.

– If the DIT station is busy and not a member of a hunt group and not forwarded, the call automatically attempt to camp-on the DIT station (voice only).

Conventional Tie (analog E&M and T1 emulating E&M)

E&M (ear and mouth) is a trunk signalling protocol where the required trunk signalling is done over separate leads. The term "E&M" in this part refers to both analog E&M and T1 emulating E&M.

1. Digits received over private analog tie trunks may be optionally prefixed with up to 6 digits for uniformity with dialed digits (e.g. access code could be provided). Prefix digits may be assigned for a non-ISDN tie trunk (DTMF signaling only).
2. Tie trunk networks use a 4-wire E&M trunk board or the T1 board, emulating E&M type wink start, immediate, or delaydial.
3. Analog tie networks consist entirely of 4-wire E&M trunk boards.
4. Digital tie networks consist entirely of TMDN boards.
5. Analog/digital combined networks use both types of boards.
6. 2-wire E&M trunks are not supported.

Flexible Intercept

Handling of external (trunk) calls which encounter error or trouble conditions that prevent normal processing (e.g., stations out of service, vacant numbers, etc.) is known as "intercept" handling.

1. Intercept handling for Direct Inward Dialing (DID) trunk calls:

   a) Called station busy or has Do-Not-Disturb feature activated: follow normal rules of forwarding, hunting, and camp-on (no intercept option). If none of these options applies, the caller has receive busy tone.

   b) Called station does not answer: follow normal rules of forwarding, hunting, and camp-on (no intercept option). If none of these options applies, the caller has receive ringback tone until he disconnects.

   c) Vacant number or out of service station/personal attendant position: intercept call to intercept announcer, if one is equipped for the Internal Traffic Relations (ITR) group in which the trunk is a member. If no announcer is assigned to the ITR group, intercept to the general attendant queue based on the ITR assigned to the incoming trunk for vacant numbers and out of service personal attendant positions, or based on the ITR assigned to the destination party for out of service stations.

   d) Incomplete dialing: reorder tone

   e) Called station blocked by classmarks (i.e., DID not allowed) or ITR group restriction: intercept to general attendant queue based on the ITR assigned to the destination party.

2. An announcement device be supported for intercept announcer operation. This type of announcer operates in the following manner:

   a) When a call requires the intercept announcement, the announcement port is alerted (rung).

   b) The announcer responds by going off-hook when ready to start the announcement.

   c) The caller hears ringback tone until the announcer goes off hook.

   d) When the announcer goes off hook (answers), answer supervision is returned to the calling (trunk) party.

   e) When the announcement is complete, the announcer signals completion by going on-hook. At this time, the calling (trunk) party is disconnected.

   f) Multiple calls can listen to the announcement simultaneously. When the announcer answers the alerting signal, all waiting calls are connected to the announcer.

   g) If new calls arrive at the announcer while the announcement is in progress, they are queued and receive ringback tone until the start of the next announcement cycle. When the announcer goes on hook, the old calls are disconnected, and the announcer is alerted (rung) again. When it goes off hook to signal the next announcement cycle, the next group of waiting calls are connected.

3. Under night service conditions, intercepted/redirected calls to the general attendant queue go to the night answer position.

4. Intercept for trunk calls to vacant number or OOS station is supported

5. Intercept on incomplete dialing supports:

   a) E&M configured for GDTR/DNIS:

   b) E&M configured for normal operation or DID translation:

6. Intercept on busy is supported

7. Intercept on not authorized (ITR restriction) or originate only is supported

8. Intercept because of DID blocking is supported

9. Intercept on all trunks busy is supported

10. Attendant intercept on do not disturb is supported

11. Intercept on no answer is supported

Aministration

The system can support one announcement device for each Internal Traffic Relations (ITR) group to handle DID intercept treatment. The same intercept announcer can be used for multiple ITR groups.

Hoot and Holler (H n H)

Hoot and Holler is an application where a device can dial into a busy nailed connection (e.g., paging equipment) and get connected through immediately. The user accessing the (trunk) device have a programmed key or access code for direct trunk access then enters a line number and hotlines/ringsdown to the destination.

Hoot and holler is realized for both the TMDN (T1) and TMEMUS (analog E&M) interface.

10.27.5 DID, SID/ANI, DNIS, Digit Conversation

Direct Inward Dialing (DID) Flexible Station Numbering (DID Digit Translation)

This feature permits the system to replace the incoming DID number by an internal number. The DID Digit Translation functionality could also be provided by DNIS Routing. This feature and DNIS SID/ANI routing are mutually exclusive.

Dialed Number Identification Service (DNIS). DNIS Routing on DNIS

This feature allows call routing and call displays based on the original called number. This number is the called party number for Trunks using ISDN protocols not supporting supplementary service -DNIS (SS-DNIS) and analog Trunks. Trunks which support the SS-DNIS (Cornet-N/NQ) use the external DNIS number provided by SS-DNIS. This feature is applicable for all trunk types.

Calling Number Identification (SID / ANI). DNIS Routing on SID/ANI

This feature allows call routing and call displays based on the calling party number (SID/ANI). This feature is applicable for all trunk types which support delivery of SID/ANI information, which is currently only AT&T4ESS. Automatic Number Identification (ANI) is a public network customer billing number stored at the public network central office serving the originator of the call. Station Identification (SID) is the calling party number of the station originating the call. The ANI may be delivered when the SID number is not available.

Direct Inward Dialing (DID) Flexible Station Numbering (DID Digit Translation).

- The parameters for DID digit translation are defined for each DID trunk individually.
- The system supports a maximum of 20 DID translation tables with 100 translations (digit conversions) each. Each DID table also supports 4 generic incoming services, (INWATS, VPN, SDS, and 900) each of which supports 100 SID/ANI preference entries.
- The system supports a maximum of 16 Service Rule Tables.

- On DID trunks the system can receive 2 to 16 digits from the Central Office. It is possible to delete none, one, or up to 14 leading digits of the received number insuring a minimum of 2 digits remaining. The system uses the leading 2 digits of the remaining digits for digit conversion as follows: Exactly 2 digits remaining are used as the index into a table to retrieve an extension number or feature access code. If greater than 2 digits are remaining, the leading 2 digits are removed from the digit string and are used as the index into a table to retrieve the prefix digits which are prefixed to the remaining digit string. The resulting extension number or feature access code is used by the system to route the call.
- DID digit translation may also be configured for E&M and PRI trunk groups.
- DID calls can be directed to local stations, private network stations, attendants, hunt groups, mailboxes (e.g., PhoneMail), and automatic answering unit/telephone answering systems.
- DID calls can be blocked by Internal Traffic Restrictions (ITR) group assignment.
- Stations can be blocked from receiving DID trunk calls.
- Intercept/Redirection of DID trunk calls (examples):

  1. Called station busy or has Do-Not-Disturb feature activated:
  When forwarding, hunting, or campon is configured, the call follows the normal rules of forwarding, hunting, and campon. Otherwise,. the call has to be intercepted, e.g., to an attendant or to busy tone.

  2. Called station does not answer:
  The call follows the normal rules of forwarding, hunting, or campon. If none of these options apply, the caller can be optionally intercepted, or the caller can receive ringback tone until the caller disconnects.

  3. Vacant number or out of service station/personal attendant position:
  The call is intercepted to an intercept announcement. If no intercept announcement is configured, the call is intercepted to the general attendant queue.

  4. Incomplete dialing: Return reorder tone to the caller.

DNIS Digit Conversion, Routing

- DID - DNIS Conversion:
  For incoming calls that arrive on a DID trunk group, where the number is part of a congruous set of numbers based on the original called number, the public network called number is converted to an internal DNIS number, local number (non-DNIS), or private network number (non-DNIS) using the normal DID conversion process.
- DNIS Trunking:
  For incoming calls that arrive on special DNIS trunk groups, where the called number is part of an incongruous set of numbers based on the original called

number, the GDTR translation table is used to convert the number to an internal DNIS number, to a local non-DNIS number, or to a private network number (non-DNIS).

- Private Network - DNIS Conversion:
  For calls that arrive via private network trunk groups which are configured to use the GDTR translation table, the private network number, the external DNIS number, or the locally generated number is converted into a number which is used to route the call. An external DNIS number can only be received from a CorNet-N/NQ trunk.
- When a number is converted to an internal DNIS number, it has been identified as a called number that requires DNIS treatment. This treatment involves routing to the true destination, optional calling number routing, and special displays based on the called number.
- If digit conversion is attempted via table GDTR and no match on the received incongruous number is found, the call has to be given intercept treatment.
- Once an internal DNIS number has been identified, the initial DNIS text display information retained throughout the life of the call unless it is overwritten.
- During the life of a call, the initial DNIS information may be overwritten one or more times by new DNIS information.
- The DNIS text display is an optional 24 character (maximum) alphanumeric display.
- The DNIS text display (DNIS text string) is provided on the second line of a two line display unit. The first line is the Visual Source ID display (SID/ANI text string, or calling party information, or trunk group information).
- During the life of a call, the initial DNIS information may be overwritten one or more times by new DNIS information.
- Attendants do NOT receive DNIS displays.
- The target number for non-ACD numbers may not be to a data device.

SID/ANI Digit Conversion, Routing, and Text Displays

- Based on the called (DNIS) number, the calling number may be used to apply special SID/ANI treatment. This treatment involves routing to the true destination and special displays based on the calling number. For calls terminating locally to ACD, the special treatment involves the assignment of the primary and overflow priority, the audible source ID, and call limit checks.
- Once SID/ANI treatment has been identified, this initial SID/ANI information is retained throughout the life of the call unless it is overwritten..
- During the life of a call, the initial SID/ANI information may be overwritten one or more times by new SID/ANI information.

- If SID/ANI routing is attempted and no match on the SID/ANI number is found the call attempts to route to the DNIS destination

10.27.6 LCR, Features Negiciation

Digit String Restriction Table

There are 0 to 15 possible restriction tables which shall block calls to certain dialed numbers. Each table entry contains a dialed number or portion of a dialed number (i.e., most sig-nificant digits) as a digit string, for example, that can accommodate Area code/office codes com-binations.

A route element may optionally be assigned to multiple restriction tables (powerset) that block certain combinations from user access.

Each table shall have the capability to store one first digit string and up to 20 sub-digit strings. The first digit string may be 1-4 digits in length, the sub-digit strings may be up to 4-digits in length. The sub-digit strings may represent office codes, for example.

Forced Authorisation Code (FAC/PIN)

An LCR voice Class of Service shall have a FAC/PIN restriction classmark which is used to in-dicate whether the station user must input an authorization code, if required by the LDPLN digit string, before the call can be completed. The FAC/ PIN can be pre-dialed and subject to the au-thorization code timer or shall be prompted for by LCR when required for the particular digit string. If the FAC/PIN does not pass verification, the call is not allowed and error tone provided to the caller.

This feature does not apply to data calls.

Associated with each LDPLN digit string shall be a FAC/PIN restriction flag used to indicate whether this destination requires a FAC/PIN code to be input prior to call completion.

Determination of forced authorization code required is based on the calling user's LCR COS and the LDPLN digit pattern. If a FAC/PIN code is required for the dialed number and the user, the system will prompt the user (via recall dial tone) for the FAC/PIN after the complete external number has been dialed.

The station user will not be forced to enter an Authorization Access Code by LCR, if the user has already entered the authorization code before the LCR access code is dialed.

This feature is a local-only feature or a network-wide fea-ture. Therefore, it use HiPath 4000 baseload's current implementation of network-wide PIN and incorporate the LCR forced authorization code feature.

The type of verification of Forced authorization via LCR is dependent on the checks for "checked", "unchecked" and "restricted":

Unchecked Forced Authorization Code: This type of verifivation is the same as the existing IM PKZ functionalty.

Checked Forced Authorization Code: This type of verifivation is the same as the existing IM mobile functionalty.

Restricted Forced Authorization Code: This type of verifivation is the same as the existing IM mobile functionalty only from the home station.

Queuing

1. Route advanced queuing, also called multi-group queuing. During the LCR route selection process, when the first local trunk group in the designated route is busy or OOS, it shall be internally queued for this call. Route selection shall advance to the next trunk group in the route and if it is busy or OOS, the call shall be internally queued for that local trunk group. This processing occurs for each trunk group until an available trunk is found or the route list is exhausted. Should any of the queued trunks free up prior to finding an available trunk, the least cost route (ascending order first to last) shall be used.

2. Off hook (standby) trunk queuing is permitted at an originating Hipath 4000 or tandem/gateway HiPath 4000 system when the user and trunks are administered as allowed (default=not allowed). This queueing is not to be confused with route advance queueing (see above). Off-hook trunk queue-ing occurs when all route elements are busy/OOS. Off-hook queueing is automatic (no user in-vocation) while the user remains off-hook. No queue tone shall be provided, the user must know that they should wait off-hook.

Restrictions:
Features/functions which are not realized are

- The FAC/PIN feature is not implemented for Functional Devices. So the feature is not implemented for an Anate switched on an a/b-adapter behind an Optiset/Optipoint.
- On-hook queueing for trunks
- Retry table for CorNet trunks during internal queueing

IMPORTANT: Since the Hipath 4000 base code has a robust LCR, Numbering Plan and Dial Plan product offering, only those functions required to address the US-specific market be implemented. Existing HiPath 4000 functionality be used where ever possible

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10.27.7 Transmission and Loss Plan

HiPath 4000 hardware fullfill transmission requirements. U.S. transmission quality requirements are specified in American National Standard ANSI/EIA/TIA-464-B, "Requirements for Private Branch Exchange (PBX) Switching Equipment" /3/ and in FCC Rules and Regulations, Part 68, "Connection of Terminal Equipment to the Telephone Network" /7/. Examples of quality requirements are frequency response, return loss, idle-channel noise, cross-talk, and quantization distortion.

Tone Level

Tone level for all tones, except "Called Party" and "Maintenance (test tone)" are given in ANSI/EIA/ TIA-464-B /3/.

"Called Party" requirements are given in American National Standard ANSI/ EIA/ TIA-496-A-1989, "Interface Between Data-Circuit Terminating Equipment and the Public Telephone Network" /6/.

Handling of tones is simplified by classifying tones having compatible level requirements into groups as shown in Table 1.

Tone level requirements, as measured at HiPath 4000 peripheral board circuit interfaces are given in Table 2. Note that tones applied to the DLI are 3 dB higher than those applied to the ONS because the DLI devices (e.g., Attendant Consoles) have a built-in 3 dB loss within its received path.

Table 1: Tone Groups having Compatible Level Requirements

a. Tone Group 1	
Tone Name	**Frequencies (see note)**
1. Dial	350 + 440
2. Recall Dial	350 + 440
3. Special Dial	350 + 440
4. Message Waiting Dial	350 + 440
5. Confirmation	350 + 440
6. LCR Route Advance	350 + 440

b. Tone Group 2	
Tone Name	**Frequencies (see note)**
1. Reorder	480 + 620
2. Busy	480 + 620
3. Audible Ringing	440 + 480
4. Special Audible Ringing	440 + 480, 440
5. Intercept	440, 620
6. On-Hook Queue	480 + 620
7. Called Party	2100

f. Tone Group 6	
1. SIU-generated Announcements	Voice-band (pre-recorded voice announcements)
g. Tone Group 7	
1. SIU-generated Music	Voice-band (synthesized music)

Note: Tone frequencies are listed here for reference purposes, only.

Table 2 : Tone Level Requirements (dB)

	ONS	OPS	DLI & ICS	A/TT	A/CO	A/TO
TONE GROUP 1						
1. 2,3,4, (Dial), 5 &6 (combined) (see note)	-13 to -23	-13 to -23	-10 to -20	-13 to -23	-13 to -14.5	-13 to -14.5
TONE GROUP 2						
1, 2, & 6 (Busy, etc.) (per frequency)	-19.5 to -35	-19.5 to -35	-16.5 to -32	-19.5 to -35	-21 ± 1.5	-21 ± 1.5
3, & 4 (Ringing, etc.) (combined)	-11.5 to -27	-11.5 to -27	-8.5 to -24	-11.5 to -27	-13 ± 1.5	-13 ± 1.5
5. (Intercept) (per frequency)	-12.5 to -33	-12.5 to -33	-8.5 to -30	-12.5 to -33	-12.5 to -15.5	-12.5 to -15.5
7. (Called Party)	-16	-16	-13	NA	NA	NA
TONE GROUP 3						
1, 2, 3, & 4 (Call Waiting)	-12.5 to -33	-12.5 to -33	-8.5 to -30	-12.5 to -33	-14 ± 1.5	-14 ± 1.5
TONE GROUP 4						
1. (Test Tone)	-17	-17	-14	-17	-17	-17
TONE GROUP 5						
1. DTMF (combined)	-1 to -3	-1 to -3	-3 nom.	-1 to -3	-1 to -3	-1 to -3

	S/ATT	S/DTT	ISD/TT, S/CTT DAL, IST & VOP	D/CO	D/TO
TONE GROUP 1					
1, 2,3,4. (Dial), 5 & 6 (combined) (see note)	-13 to -23	-13 to -23	-13 to -23	-13 to -14.5	-13 to -14.5
TONE GROUP 2					
1, 2, & 6 (Busy, etc.) (per frequency)	-19.5 to -35	-19.5 to -35	-19.5 to -35	-21 ± 1.5	-21 ± 1.5
3, & 4 (Ringing, etc.) (combined)	-11.5 to -27	-11.5 to -27	-11.5 to -27	-13 ± 1.5	-13 ± 1.5
5. (Intercept) (per frequency)	-12.5 to -33	-12.5 to -33	-12.5 to -33	-12.5 to -15.5	-12.5 to -15.5
7. (Called Party)	NA	NA	NA	NA	NA
TONE GROUP 3					
1, 2, 3, & 4 (Call Waiting)	-12.5 to -33	-12.5 to -33	-12.5 to -33	-14 ± 1.5	-14 ± 1.5
TONE GROUP 4					
1. (Test Tone)	-17	-17	-17	-17	-17
TONE GROUP 5					
1. DTMF (combined)	-1 to -3	-1 to -3	-1 to -3	-1 to -3	-1 to -3

	ONS	OPS	DLI & ICS	A/TT	A/CO	A/TO
TONE GROUP 6						
1. SIU-generated Announcements	-13 to -23	-13 to -23	-10 to -20	-13 to -23	-13 to -23	-13 to -23
TONE GROUP 7						
1. SIU-generated Music	-13 to -27	-13 to -27	-10 to -23	-13 to -27	-13 to -27	-13 to -27

	S/ATT	S/DTT	ISD/TT, SCTT DAL, IST & VOP	D/CO	D/TO
TONE GROUP 6					
1. SIU-generated Announcements	-13 to -23	-13 to -23	-13 to -23	-13 to -23	-13 to -23
TONE GROUP 7					
1. SIU-generated Music	-13 to -27	-13 to -27	-13 to -27	-13 to -27	-13 to -27

1. ANSI TIA/EIA-464-B /3/ defines tone levels on a per frequency basis. The combination of the two frequencies increases signal level by 3 dB as shown in this table.

2. Test tone level requirements shown in this table are for dial-up access to the industry standard "milliwatt test tone". Internal diagnostic applications (e.g., Telephony Diagnostic System) may use this same test tone at other (nonstandard) levels.

Design Conformance with Standards

HiPath 4000 will meet all ANSI/EIA/TIA-464-B-1996 /3/ requirements for nominal loss values within permissible variation and with the following clarification that permissible variation from ICS nominal values is derived from ANSI/EIA/TIA-579-A /5/ TOLR and ROLR range tolerances.

The following deviation from nominal levels are noted:

– The T/R level for A/CO to ICS and Low Loss A/CO to ICS is -4/-3 instead of the required -6/-3. This falls within the permissible variation from ICS nominal values as derived from ICS ROLR and TOLR range tolerances defined in ANSI/EIA/TIA-579-A /5/.

– The T/R levels for a A/TT, A/CO, A/TO and S/ATT attached to a small conference are 0/0 instead of the required 3/-3. Similarly, the T/R levels for a A/TT, A/CO, A/TO and S/ATT attached to a large conference are 0/6 instead of the required 3/3. The objective to maintain all speakers' input to the conference at the same level for summing is not mandated by standards; this variation from the objective nominal level is considered to be acceptable.

10.27.8 PRI Redirecting Number (IBM)

The IBM Redirecting Number feature is supported for any calling party device type, including (but not limited to) an incoming public/private network trunk, any station type (anate, digite, mobile, teleworker) or attendant in the local node, or any station type or attendant in another node of the private network.

The original called party's device can be any existing HiPath 4000 subscriber device that can be forwarded to a PhoneMail/Saturn messaging server.

IBM ISDN-PRI Redirecting Number (IBM-RN) provides support for sending of the ISDN-PRI Redirecting Number (RN) of the original called party for 0# transfers out of a PhoneMail or the new messaging server (codenamed "Saturn") to a public network ISDN-PRI trunk that supports the sending of Redirecting Number (RN) information to the public network.

Applicable Call Scenarios

See Figure 1. There are five different scenarios based on configuration of the private network that must support the sending of the IBM RN. The differences between these scenarios are due to the relative locations of the incoming trunk or originating station, original called party, PhoneMail/Saturn messaging server (designated as "PM" in the figure), CorNet-N/NQ trunks (if any), and the outgoing public network ISDN-PRI trunk, as follows:

Scenario A:

A call originated by a station or incoming trunk is directed to "Station A" located in the same switch. Station A is call forwarded (any type) to a PhoneMail/Saturn messaging server that is also located in the same switch. The caller opts to transfer out of PhoneMail/Saturn using the 0# transfer function, which causes the PhoneMail/Saturn messaging server to transfer the call to an off-net destination address. The transferred call is routed via an outgoing public network ISDN-PRI trunk configured for "IBM RN" located in the same switch. The SETUP message sent to the outgoing public network ISDN-PRI trunk are include an RN IE or DIV-LEG2 operation containing the explicit private network number of Station A.

Scenario B:

A call originated by a station or incoming trunk is routed via a private network trunk to "Station B" located in a different switch. Station B is call forwarded (any type) to a PhoneMail/Saturn messaging server that is located in the same switch as Station B. The caller opts to transfer out of PhoneMail/Saturn using the 0# transfer function, which causes the PhoneMail/Saturn messaging server to transfer the call to an off-net destination address. The transferred call is routed via an outgoing public network ISDN-PRI trunk configured for "IBM RN" located in the same switch as the PhoneMail/Saturn messaging server. The SETUP message sent to the outgoing public network ISDN-PRI trunk are include an RN IE or DIV-LEG2 operation containing the explicit private network number of Station B.

Scenario C:

A call originated by a station or incoming trunk is directed to "Station C" located in the same switch. Station C is call forwarded (any type) to a PhoneMail/Saturn messaging server that is also located in the same switch. The caller opts to transfer out of PhoneMail/Saturn using the 0# transfer function, which causes the PhoneMail/Saturn messaging server to transfer the call to an off-net destination address. The transferred call is routed via a private network trunk to another switch, which then routes the call to an outgoing public network ISDN-PRI trunk configured for "IBM RN". The SETUP message sent to the outgoing public network ISDN-PRI trunk are include an RN IE or DIV-LEG2 operation containing the explicit private network number of Station C.

Scenario D:

A call originated by a station or incoming trunk is directed to "Station D" located in the same switch. Station D is call forwarded (any type) to a PhoneMail/Saturn messaging server that is located in a different switch, so the forwarded call is routed via a private network trunk. The caller opts to transfer out of PhoneMail/Saturn using the 0# transfer function, which causes the PhoneMail/Saturn messaging server to transfer the call to an off-net destination address. The transferred call is routed via an outgoing public network ISDN-PRI trunk configured for "IBM RN" located in the same switch as the PhoneMail/Saturn messaging server. The SETUP message sent to the outgoing public network ISDN-PRI trunk are include an RN IE or DIV-LEG2 operation containing the explicit private network number of Station D.

Scenario E:

A call originated by a station or incoming trunk is routed via a private network trunk to "Station E" located in a different switch. Station E is call forwarded (any type) to a PhoneMail/Saturn messaging server that is located in a different switch, so the forwarded call is routed via a private network trunk. The caller opts to transfer out of PhoneMail/Saturn using the 0# transfer function, which causes the PhoneMail/Saturn messaging server to transfer the call to an off-net destination address. The transferred call is routed via a private network trunk to another switch, which then routes the call to an outgoing public network ISDN-PRI trunk configured for "IBM RN". The SETUP message sent to the outgoing public network ISDN-PRI trunk are include an RN IE or DIV-LEG2 operation containing the explicit private network number of Station E.

Note: Scenario E as shown in Figure 1 shows the PhoneMail/Saturn messaging server to be located in the same switch as the caller's station or trunk, and the outgoing public network ISDN-PRI trunk to be located in the same switch as

Station E. In reality the caller's station or trunk, Station E, the PhoneMail/Saturn messaging server, and the outgoing ISDN-PRI trunk is located in four separate switches.

Restrictions:

Interaction with other features

The following list identifies the main feature interactions.

– Caller ID Blocking / Display Suppression / Secret Station Number
– Multiple-hop Call Forwarding
– Redirecting Number Modification/Conversion
– Multimedia Messaging Services

Supported devices

IBM-RN is supported for "0#" transfers out of PhoneMail and Saturn messaging servers.

10.27.9 NI-2 PRI Calling Name Delivery

NI2-CNAM (National ISDN 2 - Calling Name Delivery) provides for the delivery of the name associated with the calling party when a call is received from the public network on an NI-2 PRI trunk (National ISDN 2 - Primary Rate Interface trunk). The calling name information is then delivered by the HiPath 4000 system to the terminating end-user device (e.g., via a display) and to Workstation Protocol (WSP) / Applications Connectivity Link (ACL) applications.

NI-2 PRI Calling Name Delivery (NI2-CNAM) affects a called enterprise system only if the enterprise customer subscribes to delivery of calling name information from the public network carrier. NI2-CNAM provides for the delivery of the name associated with the calling party when the call is delivered on an NI-2 PRI trunk. To make the delivery of calling name information meaningful for its users, the enterprise system make the information available to the end-user device (e.g., via a display) and to Siemens' Workstation Protocol (WSP) and Applications Connectivity Link (ACL) applications. This includes providing an appropriate indication when the actual name information cannot be delivered because the calling name is marked as presentation restricted or unavailable. Telcordia Technologies (formerly Bellcore) specifications exist which describe the D-Channel message flows.

Interaction with other features

When the callingName is received in the SETUP message then the existing CorNet-NQ implementation covers all feature interactions.

When the callingName is received delayed in a FACILITY message new feature interactions occur. A feature interaction in this case means the calling party number is received in alerting, connect or any other state while a feature is active. The following list identifies the main feature interactions.

- Repeat ID
- Preview Key
- Hold (Retrieve held NI-2 trunk call)
- Call Park (Retrieve parked NI-2 trunk call)
- Call Transfer
- Conference
- ACD Routing
- Call Forwarding
- Hunting
- Call Pickup
- Caller ID Blocking / Display Suppression / Secret Station Number
- Elapsed Time Display / Charged Time Display
- Camp-on
- Second Call
- Delayed Call Forwarding

Supported devices

Public network calling name information, when available, are displayed on the following HiPath 4000 end-user devices:

- Optiset E telephones (w/display)
- optiPoint telephones (w/display)
- Teleworking Clients (w/display)
- Attendant Console and AC-Win

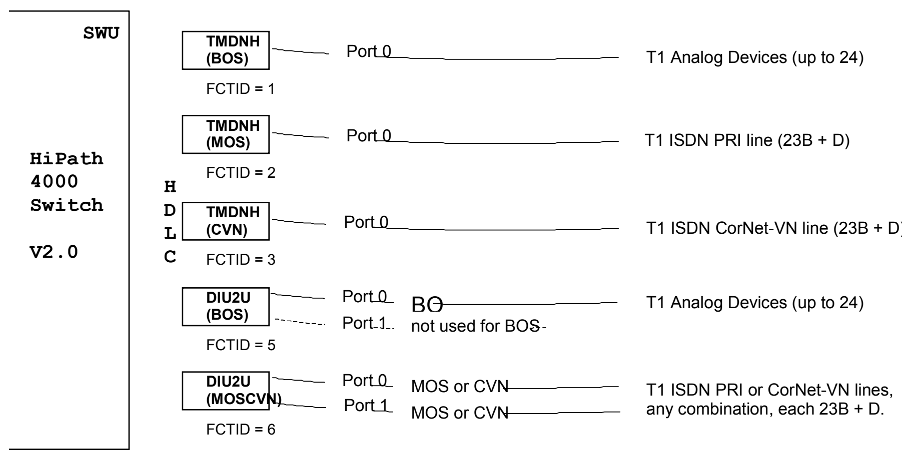
10.27.10 DIU - US

With the TMDNH interface a board with the loadware capable of supporting two T1 spans for ISDN PRI and CorNet-VN functions (MOS and CVN). Any combination of MOS and CVN can be supported by the two T1 spans.

The board can also be used for the Analog Emulation function (BOS). When used for Analog Emulation, only one span (span 0) is supported. The protocols that presently exist on the TMDNH BOS will continue to be supported.

The board's faceplate has no link LED's, only standard LED's. The T1/E1 lines attach at the front panel, not at the back plane, and a new adapter cable is required for each T1 line.

Figure 1 DIU2U with other HiPath 4000 T1 Boards



The feature TMDNH Replacement provides a board (and function IDs) to support existing TMDNH board/functions. This board and functions are the last two shown in the Figure 1 above - DIU/BOS and DIU/MOSCVN. For TMDN BOS, MOS, and CVN functions, the new board allows the TMDN to be replaced with or supplemented by the new board.

10.27.11 CorNet VN

Description:

CorNet-VN (CorNet 'Virtual Network') combines the benefits of CorNet N private networking with the ISDN-based virtual private networks (VPN) and allows CorNet VN-Customer to migrate to HiPath 4000.

CorNet-VN utilizes ISDN PRI (Primary Rate Interface) to access the public network's virtual private network service together with ISDN User-User Signaling (UUS) services to provide feature transparency over a switched public network connection. Feature transparency is accomplished by transporting CorNet-N feature control information through the switched public network using a virtual private.

There are two flavors:

One for AT&T which is used to access AT&T's Software Defined Network (SDN) and AT&T's ISDN UUS service.

The second is for MCI PRI and provides access MCI's Vnet/Vision service and MCI's ISDN UUS service N-Quest.

The CorNet-VN interface board for V6.x was the (TMDN or/an TMDN64) these boards provide the necessary interworking functions for transporting CorNet N feature control information through the public network. The interworking function, which is implemented in loadware, receives CorNet-N messages from the Hicom 300E US switching unit and maps them into a suitable for at for sending them to the AT&T or MCI public network.

10.28 Deutschland

Signalling protocoll to A6 systems

The generic functional protocol defined in ECMA-165 contains the signaling mechanism for the control of supplementary services which are not explicitly standardized by ECMA and which may be manufacturer specific. This mechanism uses call independent signaling connections (Connectionless Transport Service) and is applied in existing collocated A6-networks. The following Table 1 is the list of A6-features accordingly; these are called Additional Network Features- Generic Supplementary Services (ANF-GSS).

Hipath 4000 can be integrated in existing A6-networks; either by replacing A6-nodes by Hipath 4000-switches, or by enhancements of A6-networks with Hipath 4000.

Feature name

- Time
- ANF-CHT
- Error messages
- Traffic messaging
- Remote configurations. Examples are:
  - Subscriber configuration
  - Configuration of hunting group
  - Assignment of feature access codes
- ACD messages
- Clock synchronization
- Class of service switchover
- Activation of tracer
- PIN

- (BMO) ANF-EST
- Information system
  - Indication and retrieval of info
  - Recording of call info
  - Sending of info
- Remote control of features

Restrictions:

- Only the transit functionality is implemented in Hipath 4000 for the Connectionless Transport Service. This needs processing of the routing information in the FACILITY-message for the transit connection to be established.
- The feature transparency for all of the generic A6-services (transparent Connectionless Transport Service) is a pure networking issue and impacts neither other integrated Hipath 4000-functions nor external applications.

10.29 Japan

10.29.1 Analog DID with DTMF

The Japanese Direct Inward Dial protocol is based on the NTT national Standard. In this Standard the chapter 4.1 depicts the signaling and timing for DID in Japan. The new procedure requires an adaptation on the Pseudo DID existent in HKZ TMANI. Basically the signaling is composed by the states sequence:

- Calling signal with polarity reversal
- Primary answer signal with DC loop
- Extension designation (digits)
- Calling signal and loop disconnection
- Second Answer signal with DC Loop with reversal polarity

Specifics timings are worked here.

Previous situation

The present Loadware does not provide the functionality described on the NTT Standard. However, the states sequence in the ANMOSIG device is similar with the japanese Standard and it could be used as a basis plattform using a option possibility for the feature.

Realisation

The new timmings and states structure can be implemented in the ANMOSIG device adding a administrable parameter via PTIME. When the option is set the new sequences will be activated and it will start the J-DID protocol. No new event codes and no new messages for or from DH are foreseen in this adaptation.

Administration

The AMO PTIME shall be set in the parameter P3 (LOW Byte) as 3 to start J-DID protocol. No short or long timers must be set because all timers are fixed or used from defaul values from Iniblock PTIME 87.

Restrictions

ANMOSIG as J-DID and Normal ANMOSIG can not run simultaneously unless different Tblocks are set.

11 Appendix

11.1 Audible Tones for Germany

Tone	Frequency [Hz]	**Pulse** / Pause [ms]
Exchange dial tone	425	Continuous tone
Busy tone	425	**170**/430
ATB tone	425	**250**/250
Internal dial tone	425	**200**/300/**200**/300/**200**/800

Table 6 The above audible tones are to be recognised by the PABX

Tone	Tsl	Frequency [Hz]	**Pulse** / Pause [ms]	Level [dBm0]
Exchange dial tone	2	425	Continuous tone	-3
Internal dial tone	1	425	**200**/300/**200**/300/**200**/800	-3
Ringback tone	4	425	**1000**/4000	-3
Busy tone	5	425	**160**/440	-3
Override tone	6	425	**200**/300/**200**/1300	-14
Call waiting tone	7	425	**100**/1900	-14
Data call tone	9	1300	**600**/1800	1
Special dial tone	3	425+400	Continuous tone	-3
NU Tone	8	950/1400/ 1800	Triple tone **320-320-320**/960	-3
Conference auxiliary tone	24	425	Continuous tone	-14
LCR	26	1800	**340**/200/**340**/200/**340**/1000	-3

Table 7 The above audible tones are to be sent by the PABX

Tone	Sequence	derivated from
Conference tone	200 ms 300 ms 200 ms Cont.	Conference auxiliary tone Pause Conference auxiliary tone Silence
Call waiting tone	Cont.	Call waiting tone
Call waiting tone for night service terminal	Cont.	Call waiting tone
Override tone	Cont.	Override tone

Table 8 Special tone sequences

Type of ringing	Description	Pulse / Pause (ms)	Setting capability of pulse / pause for	
			Analog telephones	Digital telephones
Standard ringing / Internal ringing	For internal calls	**1000**/4000	Pulse 20 - 60 000 ms Pause 20 - 60 000 ms	Pulse 20 - 9 999 ms Pause 20 - 9 999 ms
Public network call / external ringing	For DID and exchange calls to the night service or forwarding station	**300**/400/ **300**/4000	Pulse 20 - 999 ms Pause 1 20 - 999 ms Pause 2 20 - 60 000 ms	Pulse 20 - 9 999 ms Pause 20 - 9 999 ms
Special ringing 1	Timed reminder, messenger call, call forwarding – don't answer, call forwarding – all calls	**1000**/1000	Pulse 20 - 60 000 ms Pause 20 - 60 000 ms	Pulse 20 - 9 999 ms Pause 20 - 9 999 ms
Special ringing 2	Direct station selection, standard multi-address call, urgent call at the AC	**300**/400/ **300**/1000	Pulse 20 - 999 ms Pause 1 20 - 999 ms Pause 2 20 - 60 000 ms	Pulse 20 - 9 999 ms Pause 20 - 9 999 ms
Special ringing 3	Triple ringing –For standard internal call and for consultation for specific users –For DGV-initiated calls if there is an appropriate classmark	**200**/200/ **200**/200/ **200**/4000	Pulse 20 - 60 000 ms Pause 20 - 60 000 ms	
Multiple tone ringing	Multiple tone ringing for ringing between executive and secretary –For Digite 260: cadence as for standard ringing, only with different pitch –For set 400:as for special ringing 2	**1000/**4000		Pulse 20 - 9 999 ms Pause 20 - 9 999 ms
Emergency ringing	Time hotline service call	Continuous ringing		

Table 9 Ringing Tones for Germany

Type of ringing	Description	Pulse / Pause (ms)	Setting capability of pulse / pause for	
			Analog telephones	Digital telephones
Timed alerting tone (AT)	If call is not accepted within a call pickup group aftertimeout. With setting: without timed AT ormultiple AT in all conditions ormultiple AT only in idle condition orsingle AT in all conditions orsingle AT only in idle condition	Ringing pulse can be set via AMO (1 - 9999 secs)	Pulse 20 - 999 secs.	Pulse 20 - 999 secs

Table 9 Ringing Tones for Germany

11.2 Assorted Overviews and Tables

	Attendant Console		DP / DTMF		optiset E		
	Code	Key	Code	Key	Code	Menu	Key
Outgoing call	x		x		x		
Incoming call	x		x		x		
Display suppression	x				x	x	
Consultation hold	x	x	x	x	x	x	x
Call transfer	x		x		x	x	x
Associated dialling			x		x		x
Call forwarding			x	x	x	x	x
Call forwarding on no answer			x		x		
Service indicator	x		x		x		
Call waiting indication (manual)			x		x	x	x
Multi-service operation							
Hold	x		x		x		x
Redial	x	x	x	x	x		x
Call tracing (manual)			x		x	x	
Three-party conference			x	x	x	x	x
Callback			x	x	x	x	x

Table 10 List of Configurable Features per Telephone Type

	Attendant Console		DP / DTMF		optiset E		
	Code	Key	Code	Key	Code	Menu	Key
Do-not-disturb			x		x	x	x
Override	x	x	x		x	x	x
Date/Time	x				x		x
Integrated executive/secretary system					x		x
Name keys (repertory keys)	x	x			x	x	x
Exclude from hunting group			x		x	x	x
Class-of-service switchover	x	x	x		x	x	
Timed reminder function			x		x	x	x
Parking	x	x			x	x	x
Voice calling / handsfree answer					x	x	x
Enable call waiting indication for second calls			x		x	x	x
Voice Mail Service (VMS)			x		x	x	
Direct station select					x	x	x
Alternating/Toggling	x	x	x		x	x	x
xx) programmable by menu							

Table 10 List of Configurable Features per Telephone Type

1. User	**S_0/S_2 CorNet-NQ (priv.)**	**S_2 DPNSS1**	**QSIG-V1 QSIG-V2 PSS1**
Consultation hold (via 2nd line)	from V 3.4	from V 3.4	from QV1
Alternating (via 2nd line)	from V 3.4	from V 3.4	from QV1
Alternating (via same line)	from V 3.4	–	–
Add-on-conference (only after consultation hold via 2nd line)	from V 3.4	from V 3.4	from QV2/P
Call forwarding (without "transportation")	from V 3.4	from V 3.4	from QV2/P
Extension of ringing (call forwarding on no answer)	from V 3.4	from V 3.4	from QV2/P
Secret call number / Suppression of display	from V 3.4	–	from QV1
Service changeover during call	from V 3.4	–	–

Table 11 Cross-System Features (Voice Features)

1. User	**S_0/S_2 CorNet-NQ (priv.)**	**S_2 DPNSS1**	**QSIG-V1 QSIG-V2 PSS1**
Paging ('multiple' paging with <u>masked</u> numbering, 'display', 'meet-me')	from V 3.4	–	–
Callback			
– on busy	from V 3.4	from V 3.4	from QV1
– on no answer	from V 3.4	–	from QV1
Third-party monitoring/witness facility (not publ. netw. subscriber)	from V 3.4	–	–
Transfer of internal call through			
– pickup	from V 3.4	–	–
– transfer before/on answer	from V 3.4	from V 3.4	from QV1
Transfer of external call through			
– pickup	from V 3.4	–	–
– transfer on answer	from V 3.4	from V 3.4	from QV2/P
Busy override camp on:			
– main PABX station to satellite	from V 3.4	from V 3.4	–/from QV2/P
– satellite PABX stations on main PABX station	from V 3.4	from V 3.4	–/from QV2/P
Display of call number/name on DIGITE	from V 3.4	from V 3.4	from QV2/P
Use of toll/code restriction of main PABX from satellite PABX	from V 3.4	–	–
Use of the central autom. dialer of the main PABX from satellite PABX	from V 3.4	–	–
Discriminating ring cadences for main PABX and SAT calls	from V 3.4	–	–
Automatic call charge registration in main PABX			
– via MFC call charge line	–	–	–
– via signaling	from V 3.4	–	–
Charge display at station terminal	from V 3.4	–	from QV2/P
Use of special facilities of main PABX or satellite PABX			

Table 11 Cross-System Features (Voice Features)

1. User	S_0/S_2 CorNet-NQ (priv.)	S_2 DPNSS1	QSIG-V1 QSIG-V2 PSS1
– Dictation equipment (DICT)	from V 3.4	–	–
– Public address system (SPKR)	from V 3.4	–	–
– Recorded announcement equipment (TCOM)	from V 3.4	–	–
– Paging system (CC)	from V 3.4	–	–
– Door opener function	from V 3.4	–	–
Multiservice operation	from V 3.4	–	–
Network-wide use of server	from V 3.4	–	–
Prevention of override/camp-on	from V 3.4	–	–
Automatic connection setup (hotline)	from V 3.4	–	–
Do-not-disturb	from V 3.4	–	–
Override do-not-disturb	from V 3.4	–	–
Flash-hook by means of VMS	from V 3.4	–	–
Central attendant console	from V 3.4	–	–
Prevention of transfer	from V 3.4	–	–
Hunting group	from V 3.4	–	–
PIN network-wide / follow-me	from V 3.4	–	–
Legend: from V 3.4: available from Hicom 300 V 3.4 from QV1: available in QSIG version 1 from QV2/P: available in QSIG version 2 or PSS1			

Table 11 Cross-System Features (Voice Features)

Country	Feature	Network-wide function with S_0/S_2 networking
Netherlands (ALS70D)	Detection of 2nd trunk dial tone; Transfer to user	no
	Tracing of incoming trunk calls (line tracing with/without register recall)	no
	Recall incoming (automatic recall on AC by means of line signal from exchange)	no
	Attendant intercept incoming without DID	yes
	ISDN	yes
Republic of South Africa (MFC-R2)	Busy PABX user override (attendant intercept after "toll exchange override" with user busy)	yes
	A-number identification for outgoing trunk calls	yes
	ISDN	yes
Finland (MFC-R2)	Busy PABX user override (attendant intercept after "toll exchange override" with user busy)	yes
	A-number identification for incoming and outgoing trunk calls	no
	Call tracing with A-number identification for incoming and outgoing trunk calls	no
	Line tracing with incoming trunk calls	no
	Line tracing with outgoing trunk calls	no
	Call forwarding to the exchange	no
	ISDN	yes
Finland (N2)	Line tracing with incoming trunk calls	no
	Line tracing with outgoing trunk calls	no

Table 12 List of country-specific features with network-wide function

Country	Feature	Network-wide function with S_0/S_2 networking
Italy (Loop / MSI)	End-of-dialing recognition with DID after fixed number of digits or time	yes
	Toll exchange forward transfer signal for incoming trunk calls by means of 25-Hz signaling from the exchange	yes
	Transmission of end-of-dialing signals (idle/busy) to the exchange with DID	yes
	Attendant offering with DID (with the two following variants):	
	• Variant 1 = Attendant intercept after "toll exchange override" with busy user	yes
	• Variant 2 = Continued B-tone application to exchange after "toll exchange override" with busy subscriber	yes
	Attendant intercept with DID and subscriber does not answer within predefined time	yes
	Add-on override by attendant	no
	Override security by the subscriber (can be prevented by service terminal)	yes
	ISDN	yes
France (Socotel / HKZ-MOSIG)	Ringback tone for waiting external parties if internal party becomes idle	no
	DID to busy subscriber (external party camps on to first-time busy internal party with DID)	yes
	Detection of 2nd trunk dial tone; Transfer to user	no
	Call waiting tone after extending a trunk call to a first-time busy user	yes
	Priority/overload display on the AC	– 2)
	Add-on conference override by the attendant	no
	ISDN	yes

Table 12 List of country-specific features with network-wide function

Country	Feature	Network-wide function with S_0/S_2 networking
France (ISDN)	Basic Call	yes
	Calling party identification (line number supplied by exchange / user)	yes
	Transmission of a subaddress (max. 4 digits)	yes
	Emergency connection (priority of outgoing connections as against incoming connections)	yes
	Direct inward dialing	yes
	Secret call number (no call number transfer of calling party to called party)	yes
	Call charge information at station during call	yes
	Call charge information at station at the end of the call	yes
	Call forwarding per terminal to the exchange	yes
	Call tracing (register call number in the exchange)	yes
Spain	Detection of 2nd trunk dial tone; Transfer to user	no
Switzerland (ISDN)	Basis Call	yes
	Direct inward dialing	yes
	Display, suppress, force, register calling user identification	yes
	Transmission of a subaddress (max. 4 digits)	yes
	Call charge information at station during the call	yes
	ISDN	yes
United Kingdom	DASS2	yes 3)
	DPNSS1	yes 3)
	ISDN	yes
Belgium	ISDN	yes
Denmark	ISDN	yes
Luxembourg	ISDN	yes
Austria	ISDN	yes
Portugal	ISDN	yes

2) This feature is not rimplemented

3) Feature according to gateway specification

Table 12 List of country-specific features with network-wide function

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