

Documentation

HiPath 4000 V5 IP Solutions, SIP Connectivity

Service Documentation

A31003-H3150-S104-2-7620

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1 Overview

In addition to AMO BFDAT the used B channels have to be specified for the selected trunking protocol in AMO CGWB. There are two options for SIP trunking (AMO CGWB):

TRPRSIP	native SIP trunking, e.g. SIP carriers with UDP or TCP In addition to this parameter, "Native SIP" or "Provider" must be set for the board in WBM (Explorers > Basic Settings > (right-click) Gateway > Edit Gateway Properties > SIP Protocolvariant for IP Networking).
TPRSIPQ	SIP trunking with with SIP using CorNet NQ (CorNet NQ over SIP with TCP), e.g. OpenScape Voice

1.1 Procedures

The „SIP Trunking“ feature uses procedures described in the ECMA 355 standard to connect HiPath systems, external server (e.g. OpenScape Xpressions) or SIP service providers via IP-based networks.

CorNet-NQ messages and CorNet-NQ features are tunneled over SIP or SIP-Q.

SIP-Q V2 supports payload negotiation, encrypted using the **SRTP** (Secure Real Time Protocol). In this case, signaling is secured using **TLS** (Transport Layer Security).

1.2 Communication Matrix

1.2.1 SIP Trunking

Legend for the following table:

BC	...	Basic call only
N/I	...	Not impacted by SIP feature

	HiPath 2000	HiPath 3000	HiPath 4000	OpenOffice EE	OpenScape Voice	HiQ8k
HiPath 4000	x	x	x	x	x	x (BC)

Table 1 SIP trunking communication matrix

HiPath 2000 V2

Networking with CorNet-IP or SIP-Q V2 (as of HiPath 2000 V2 SMR 4)

HiPath 3000 V7

HiPath 3000 V6.0 / V7 offers the same SIP and security functions as HiPath 4000.

Networking with CorNet-NQ (TDM), CorNet-IP or SIP-Q V2 (as of HiPath 3000 V7 SMR 4)

OpenScape Voice V3.1

The IP connection is implemented using SIP-Q V2.

OpenOffice EE V1

The IP connection is implemented using SIP-Q V2.

HiQ8K

Networking using native SIP. Only basic call is possible.

1.2.2 SIP Subscriber

Legend for the following table:

BC ... Only basic call

N/I ... SIP feature has no influence

	SIP subscriber	HFA subscriber	HP3k/5k H.323 standard subscriber	HiPath 3k/4k TDM subscriber	PSTN subscriber
SIP subscriber	ok	ok	ok (BC)	ok	ok (BC)

Table 2 SIP subscriber communication matrix

1.3 Boards Used

The common gateway board HG 3500 is used as of HiPath 4000 V4. For details, see "HiPath Gateways Hg 3500 and HG 3575", [Chapter 4, "Supported Gateways"](#).

2 Features and Restrictions

2.1 Features

- CLIP / CLIR
- COLP / COLR
- RFC 3261 basic call
- DTMF signaling (Inband and RFC2833)
- Name display
- Hold / Retrieve / Toggle
- Call transfer (blind, unattended, attended)
- Call forwarding (immediately, no answer and busy)
- Call deflection
- Call waiting (take, reject or forward the call)
- Second call (forward the second call, hold)
- Callback on busy subscriber / Callback No Reply
- Do Not Disturb
- Digest Authentication
- End-To-End Payload
- SIP session timer (more information can be found in [Section 2.1.1, "SIP Session Timer"](#).)
- Message Waiting Indication
- Route Optimization
- HiPath 4000 as Survivability Media Gateway
- Signaling and voice encryption (TLS / SRTP) with SIP-Q trunking
- local 3-party conference
- Hunt group (only stations, no master)
- Parallel call / ONS group
- T38 fax for
 - SIP subscriber

- native SIP trunking and
- SIP-Q trunking
- Video support at HiPath 4000 SoftGate (see „HiPath 4000 SoftGate“ > [Chapter 7, “Video Connections”](#))

All other features that are offered by the SIP subscribers but require board support do not work.

All feature events (with the exception of those referred to above) from the system's call processing are rejected.

Please refer to [Section 2.2, “Restrictions”](#) for more detailed information about restrictions.

2.1.1 SIP Session Timer

The "SIP Session Timer" provides a "keep alive" system for SIP sessions. If this feature is activated, reINVITE or UPDATE messages are sent periodically during an established connection. If the partner answers, the session is extended by a period of time previously arranged between the partners. This verifies that the connection is still established.

If the session ends without an extension, the connection is released.

The marginal values for the duration of the extension can be set in the WBM of the gateway:

WBM -> Explorer -> Voice Gateway -> SIP Parameters

SIP Parameters

RFC 3261 Timer Values	
Transaction Timeout (msec):	<input type="text" value="32000"/>
SIP Transport Protocol	
SIP via TCP:	<input type="text" value="Yes"/>
SIP via UDP:	<input checked="" type="checkbox"/>
SIP Session Timer	
RFC 4028 support:	<input checked="" type="checkbox"/>
Session Expires (sec):	<input type="text" value="1800"/>
Minimal SE (sec):	<input type="text" value="90"/>
DNS-SRV Records	
Blocking time for unreachable destination(sec):	<input type="text" value="60"/>
Outgoing Call Supervision	
MakeCallReq Timeout (sec):	<input type="text" value="3"/>

Figure 1 SIP session timer

- The SIP session timer is activated by the **Use RFC4028** checkbox.

- **Session Expires:** Determines the maximum time by which a session can be extended.
- **Minimal SE:** Determines the minimum time by which a session can be extended.

The actual duration of the extension also depends on the partner's settings but is between the configured interval. A session is renewed as soon as half duration has expired.

IMPORTANT: Please note that very short intervals lead to increased message traffic!

2.2 Restrictions

2.2.1 SIP Trunking

- SRTP is not supported for native SIP trunking. SRTP is however supported for SIP-Q trunking .
- Outband signaling is not supported.

The function must be deactivated via WBM:

WBM > Explorer > Payload > HW Modules

DSP Settings

The screenshot shows the 'DSP Settings' window with two tabs: 'General' and 'Fax Parameter'.

General Tab:

- Echo Canceller: ☒
- DTMF Outband Signaling: ☐
- Default DTMF Tone Duration (msec):
- Default DTMF Pause Duration (msec):
- Max. No. of Bytes for G.711: 960
- Max. No. of Bytes for G.723: 96
- Max. No. of Bytes for G.729: 120

Fax Parameter Tab:

- Error Correction Mode: ☐
- Number of Redundancy Packets:
- Maximum Network Jitter (hex msec):

- Trunking between HiPath 3000/5000 and HiPath 4000
SIP registration of the HG 1500 board at the Large Enterprise Gatekeeper (LEGK) is not supported.

2.2.2 SIP Subscriber

2.2.2.1 Terminal

The following entries must be deactivated on SIP terminals (either in the DLS “Feature Availability” menu or on the terminal directly), as these features are **not** supported by the HiPath 4000:

DLS	Stn.	Web Interface
DLS -> Workpoints -> optiPoint Configuration -> Feature -> Feature Availability		
Log forwarded calls	Log Forwarded Calls	Log forwarded calls
Call display name	Call Display Name	Call display by name
Music on Hold	Music On Hold	Music on hold
Message Waiting	Message Waiting	Message waiting
Auto answer - CTI	Auto Answer CTI	Auto answer - CTI
Auto reconnect - CTI	Auto Reconnect CTI	Auto reconnect - CTI
Park	Call Park	Call park
Call Park Pickup	Call Park Pickup	Call pickup
DLS -> Workpoints -> optiPoint Configuration -> Feature -> Feature Settings 2		
Callback - busy	Callback-busy	Callback - busy
Callback - no reply	Callback-no reply	Callback - no reply

Tabelle 3 Feature restrictions on the SIP endpoint

2.2.2.2 System Restrictions

The following functions are not supported in the HiPath 4000 for SIP subscribers:

- Exec. / Secr. - SIP terminals cannot be configured as these members.
- Pickup group (including Group Call) - 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))
- ACD (automatic callback (CCBS / CCNR))
- Emergency override / emergency release

- Mailbox
- Integrated call log
- Night server (only possible passively, not actively)
- Parallel ringing - SIP terminals must not be configured as members of a Mulap.
- Silent Monitoring - Silent Monitoring cannot be configured for an SIP terminal.
- Witness Facility - An SIP terminal can only be the passive subscriber for the witness facility.
- Voice calling
- 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
- ACL monitoring and related features
- 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.
- Conference - An SIP terminal can take part in a conference, but cannot initiate one.
- Display suppression
- SRTP is not supported for SIP stations.

Features and Restrictions

Restrictions

3 Direct Media Connect (DMC)

IMPORTANT: DMC in connection with HiPath 4000 SoftGate please refer to the documentation „HiPath 4000 SoftGate“ > [Section 1.2, “Restrictions”](#).

3.1 Feature Description

One major function on the LAN side is “direct payload” (DMC → Direct Media Connect) between the call parties. This function can be activated/deactivated for all SIP endpoints.

Given the fact that direct payload transmission between SIP subscribers and other IP devices and gateways is desirable (due to better voice quality), despite the fact that SIP subscribers do not support the H.323-based DMC procedure, HG 3500 has been extended with HiPath 4000 V3.0 to include a “DMC proxy”. This feature is also included in the common gateway board. In other words, as far as DMC signalling is concerned, the gateway functions on behalf of the SIP subscriber and converts the DMC signalling into SIP signalling.

For HFA subscribers the DMC connection is shown in the following diagram.

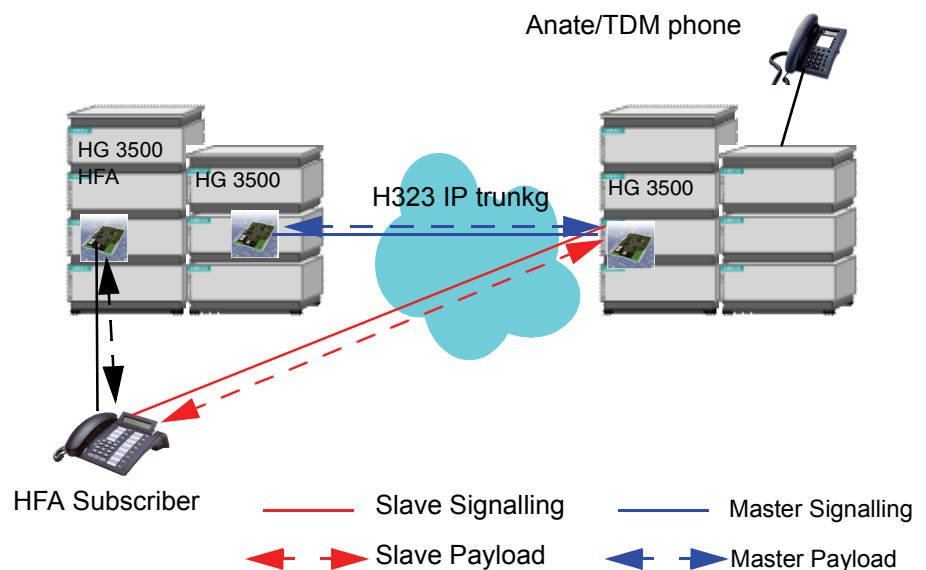


Figure 2 DMC procedure used in HiPath 4000

This figure shows the situation for an HFA subscriber calling a TDM subscriber in another node via an H323 IP trunk. The master connection for both signalling and payload is maintained via the HG 3500 (in this case configured as a HFA

Direct Media Connect (DMC)

Feature Description

gateway) and both HG 3500 (IP trunking) gateways. The Slave connections are the DMC connections and these are between IP endpoints, on the one side the HFA subscriber and on the other side the HG 3500 IP trunking board.

The transport addresses (IP addresses/ports) for payload transmission are transmitted as part of the DMC signalling. The DMC proxy does not transmit the actual transport address in this case, but instead transmits the address of the addressed SIP subscriber in order to facilitate a direct RTP stream for the SIP subscriber.

Given the fact that H.323 and SIP both use RTP for voice transmission, connection between

- H323,
- SIP and HFA subscribers,
- IPDA HiPath Host system and IPDA Access Points,
- Common Gateways (IP-/SIP-trunking) and HG 1500 gateways

is possible.

To set up a standards-compliant payload channel for the SIP subscriber, the SIP re-invite and, preferably (provided the subscribers support it), the SIP update procedures are used (the latter also requires SIP-PRACK support). These procedures make it possible to switch a payload connection over to another destination. Unlike previous DMC endpoints, the “master” payload connection to HiPath is not maintained in this case - i.e. the DMC payload connection replaces the master connection. However, the “master” signalling for the HiPath remains established.

This is illustrated in the figure below:

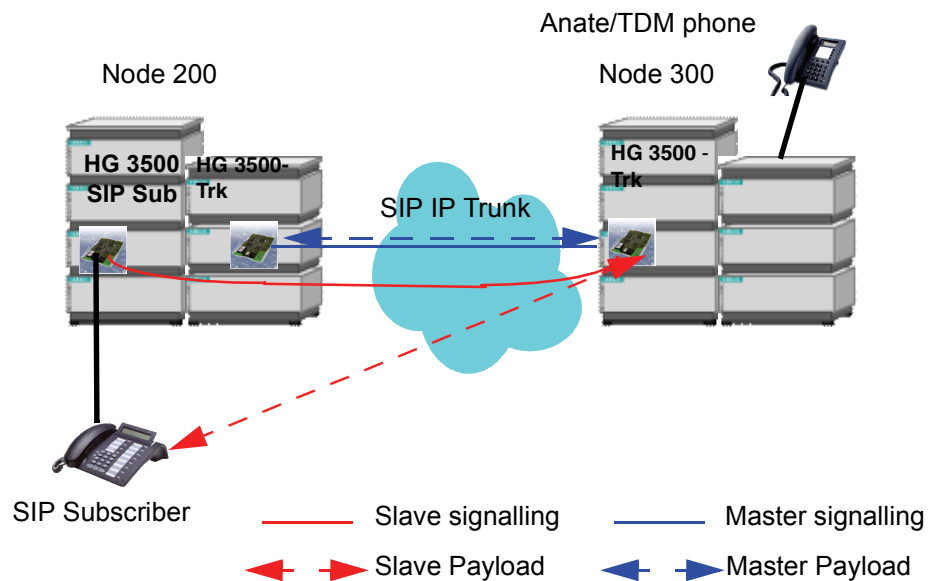


Figure 3 HG 3500 subscriber which functions as a DMC proxy

Explanation:

Figure 2 DMC procedure used in HiPath 4000: DMC signalling and payload between an HFA telephone and an HG 3500 trunking gateway (see also H323 / H323 Annex Connectivity > [Chapter 2, "Payload Switching DMC"](#)).

Figure 3 HG 3500 subscriber which functions as a DMC proxy: DMC payload is sent between the SIP phone in node 200 and an HG 3500-Trunking in Node 300. DMC signalling is between the DMC proxy in HG 3500 board and the HG 3500-SIP-Trunking board.

A distinction is made between the DMC proxy and the DMC endpoint. The latter is the endpoint for DMC signalling and the DMC payload channel (RTP) and is used as of HiPath 4000 V2.0. Both gateways (HG 3500, HG 3575) and subscribers function as DMC endpoints. HG 3500-Trk in node 300 and the SIP phone in node 200 are therefore DMC endpoints in the diagram.

Note that the HG 3500 in this example is not part of a payload connection. However, a DSP is reserved for the connection in case activation of a feature terminates the DMC and causes a fallback to the master connection. When the DMC is terminated, a master payload connection must be renegotiated by SIP re-invite, as the connection does not exist for the SIP subscriber.

Given the fact that the SIP subscribers are themselves not DMC-capable, we no longer talk about DMC connections as of HiPath 4000 V3.0, but instead more generally about an end-to-end payload connection, or e2epc for short.

Direct Media Connect (DMC)

Restriction

Setup of the master connection, which leads to a master payload connection between the calling SIP endpoint and HG 3500 (requested in order to transmit announcements and tones from the HiPath to the SIP endpoint) is not displayed. As is the case in HiPath 4000 V2.0, the DMC is initiated with the Connect of the called side.

The procedure described above is even used for a call between two SIP subscribers, i.e. an H.323-based DMC is set up from HG 3500 to HG 3500 (or internally if both subscribers are registered on the same gateway). A direct connection between HiPath 4000 SIP subscribers without DMC is not possible at present.

3.2 Restriction

IMPORTANT: When connecting a fax adapter to a HG 3500 (SIP subscriber), DMC must be deactivated for this device (only then is T.38 transmission possible).

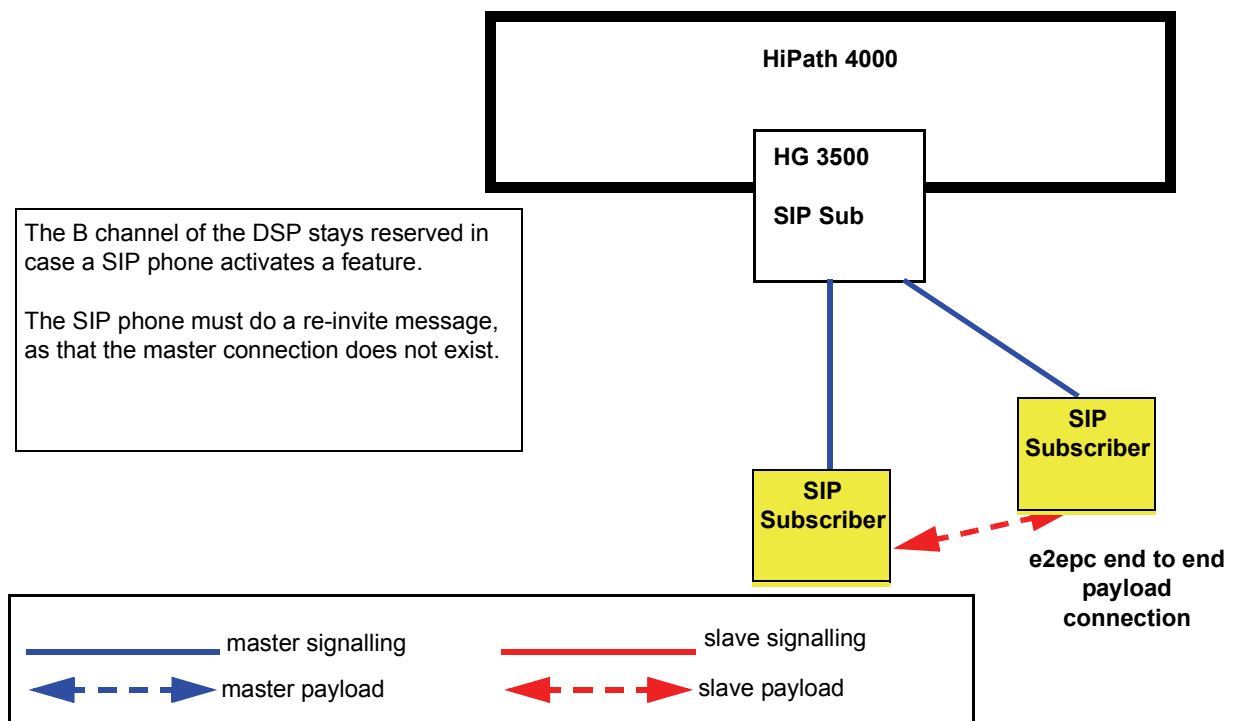
If you want nonetheless to connect DMC for this device, the codec must be set to G.711 (fax transmission with G.711, not T.38).

3.3 Scenarios

The diagrams in the following sections illustrate the possible scenarios in which a SIP subscriber is involved in an e2epc, for example DMCs to HFA and SIP stations in the same (or in a different) node, and DMCs to IPDA and IP trunking boards - the latter functioning as DMC endpoints.

3.3.1 SIP Subscriber at same HiPath 4000

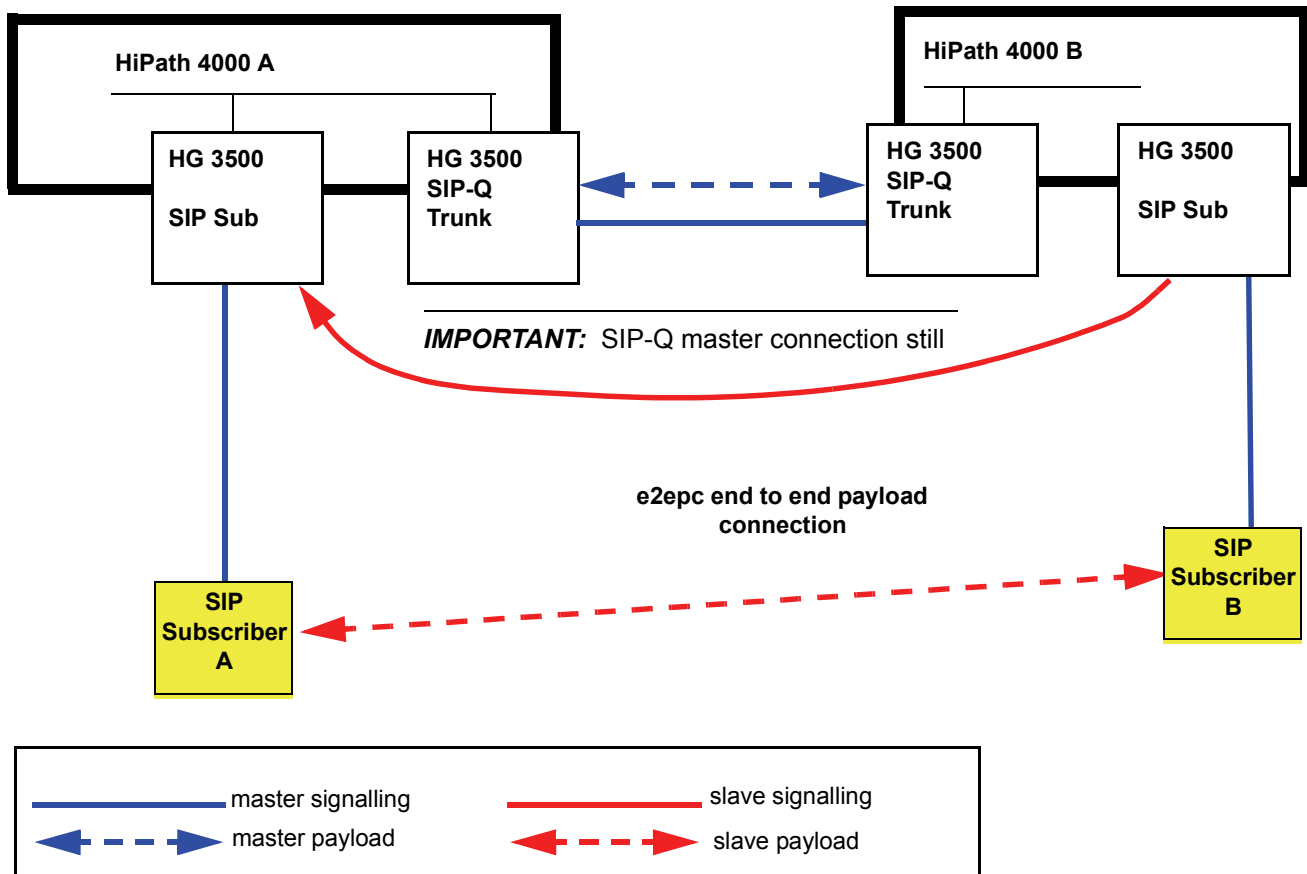
DMC signalling between the HG 3500s (or internally, if both are on the same gateway), e2epc between the stations, which replaces the master payload connection to the gateway.



3.3.2 SIP Subscriber at another HiPath 4000

Slave signalling to HG 3500 SIP subscriber board, slave payload to the SIP subscriber.

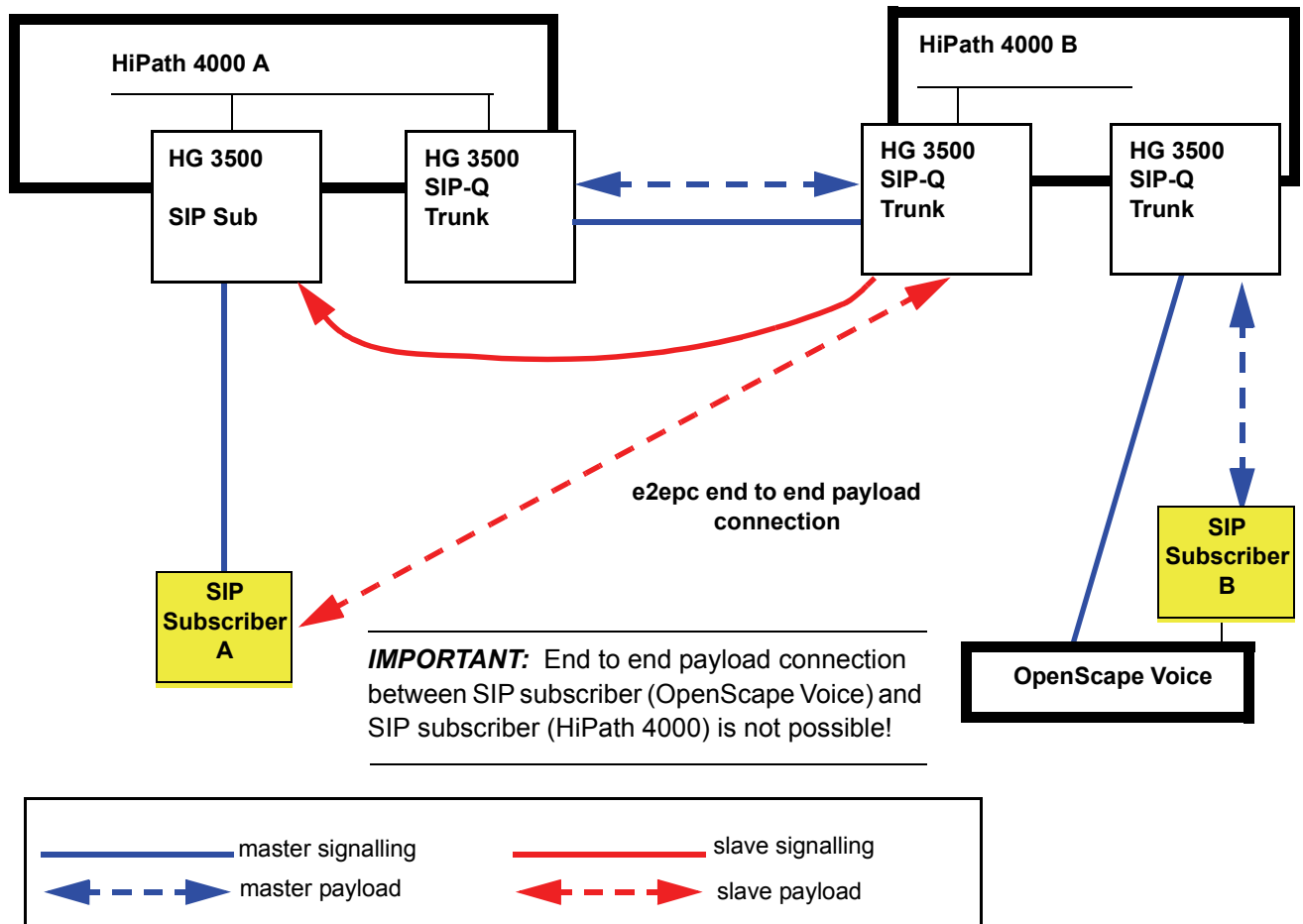
SIP subscriber A sets up a call to HiPath 4000 A. HiPath 4000 A routes the call via a trunking connection to HiPath 4000 B. HiPath 4000 B then sets up a connection to SIP subscriber B. HiPath 4000 B initiates a DMC between SIP subscriber B and SIP subscriber A, resulting in an end-to-end payload connection.



3.3.3 SIP Subscriber at OpenScape Voice

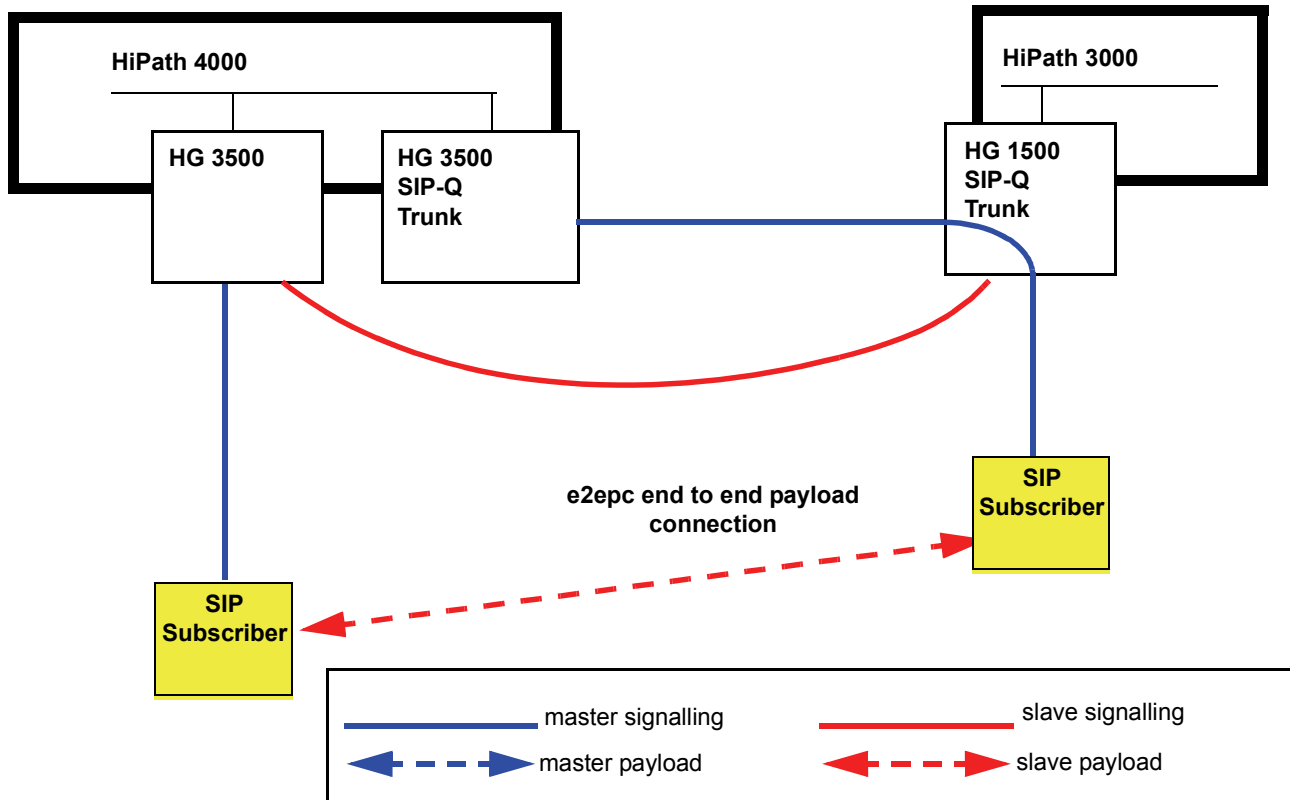
Slave signalling to HG 3500 SIP subscriber board, slave payload to the SIP subscriber.

SIP subscriber A sets up a call to HiPath 4000 A. HiPath 4000 A routes the call via a trunking connection to HiPath 4000 B. HiPath 4000 B routes the call via another trunking connection to OpenScape Voice. OpenScape Voice then sets up a connection to SIP subscriber B. HiPath 4000 B initiates a DMC between SIP trunking gateway B and SIP subscriber A, resulting in an end-to-end payload connection.



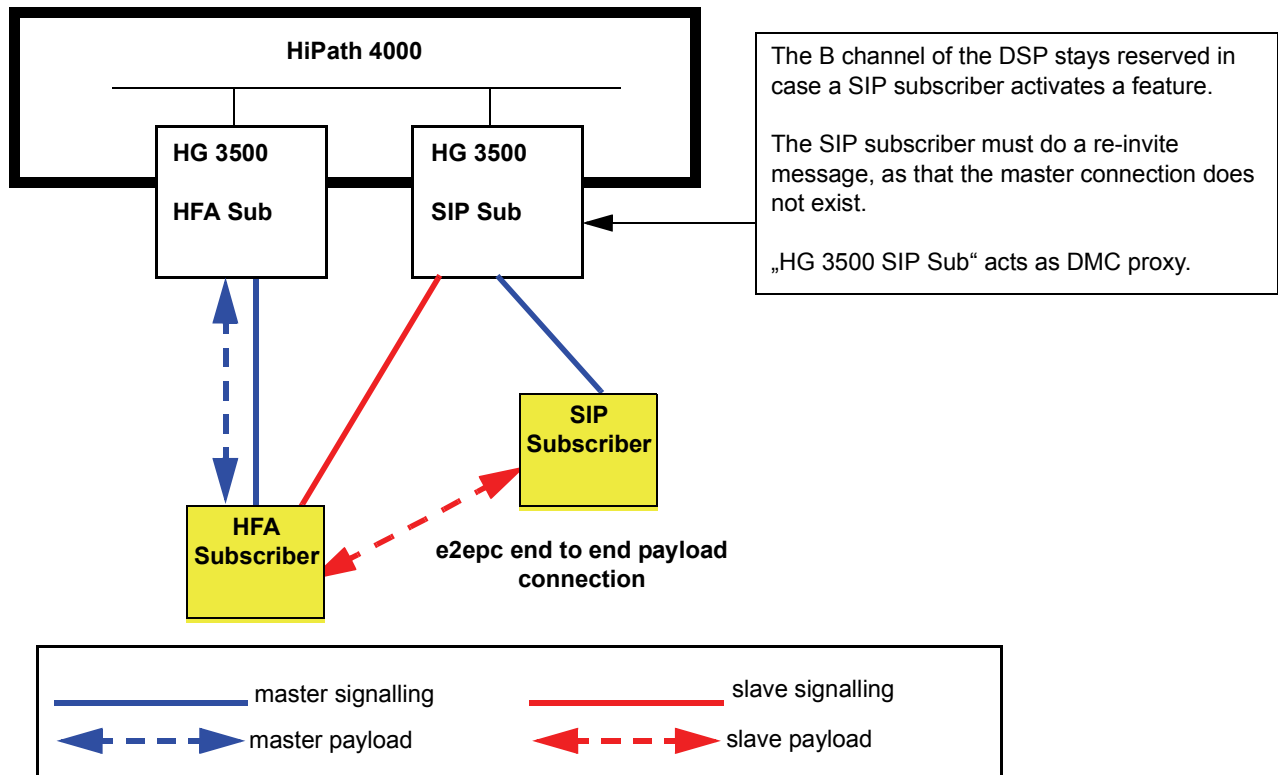
3.3.4 SIP Subscriber at HiPath 3000

DMC signalling between HG 3500 and HG 1500 (initiated by HiPath 4000), e2epc between SIP subscribers.



3.3.5 HFA Subscriber at same HiPath 4000

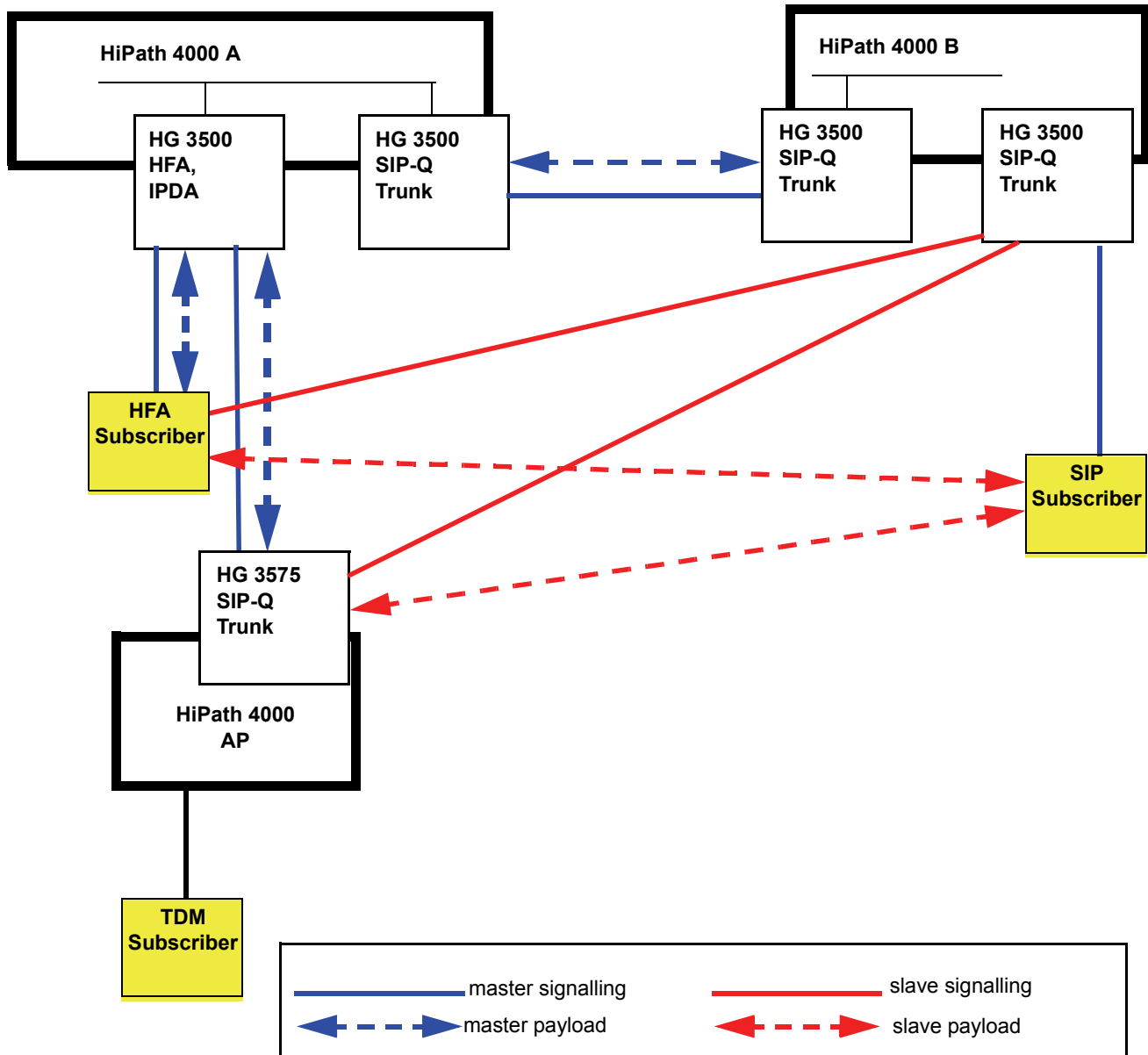
DMC signalling between HG 3500 and HFA subscriber, e2epc between the stations. A master payload connection to the HG 3500 (HFA subscriber) is retained.



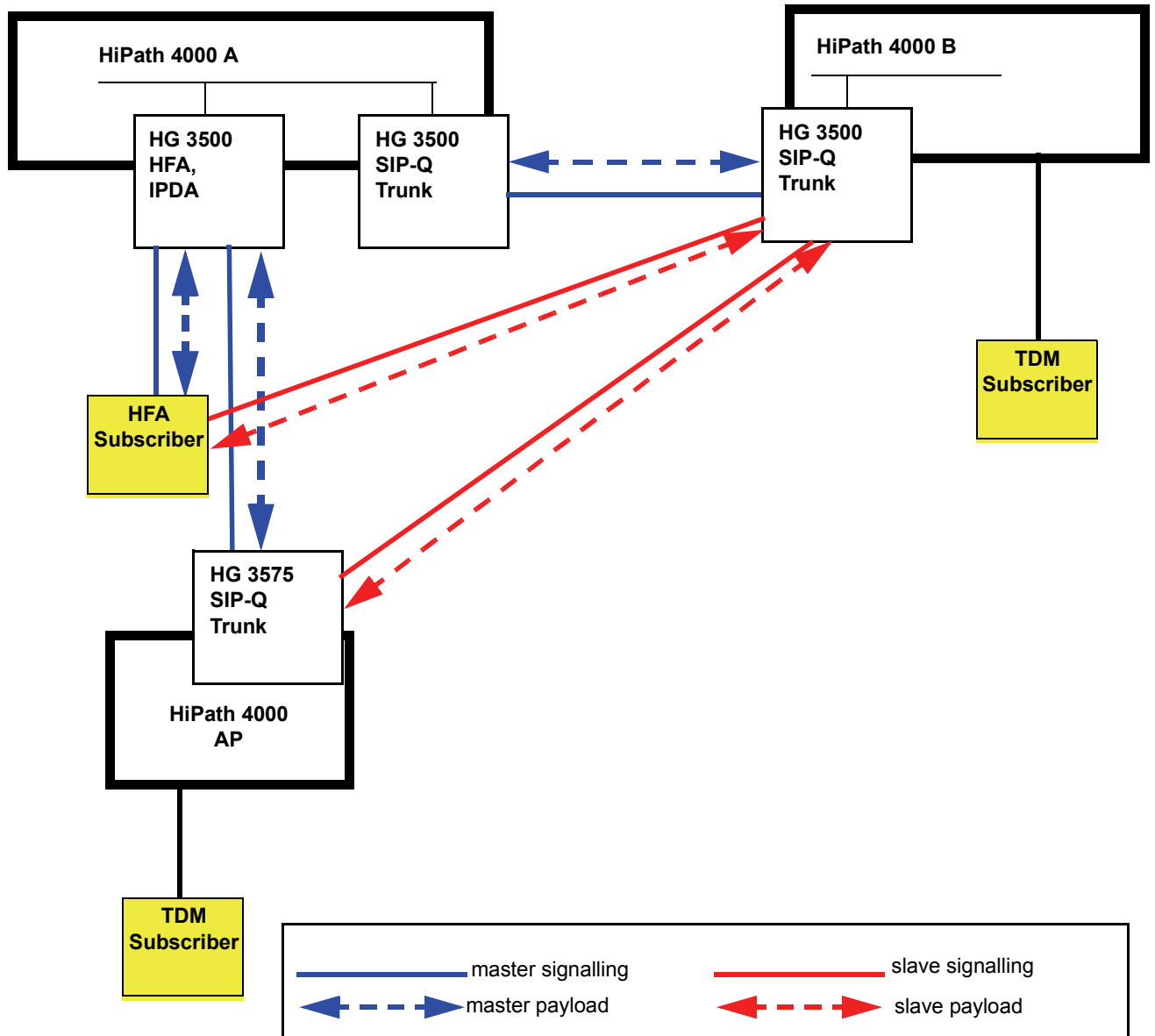
3.3.6 HFA Subscriber at another HiPath 4000

Example 1: End-to-end payload between HFA and SIP subscriber (also HG 3575 and SIP subscriber)

No specific action is necessary with the HG 3500 trunking boards. HiPath 4000 B initiates a DMC between the SIP subscriber and HFA subscriber, resulting in an end-to-end payload connection.

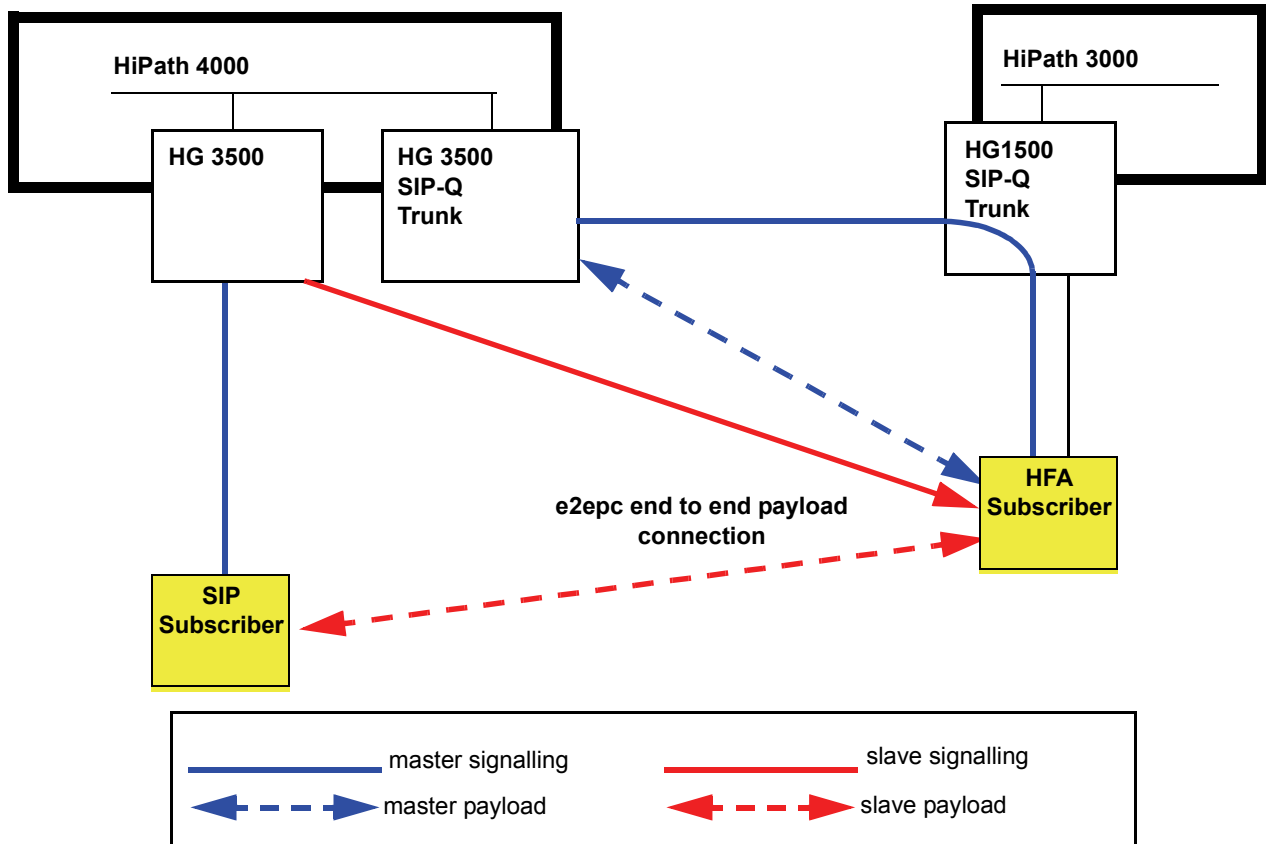


Example 2: End-to-end payload between HFA and SIP trunk (also HG 3575 and SIP trunk)



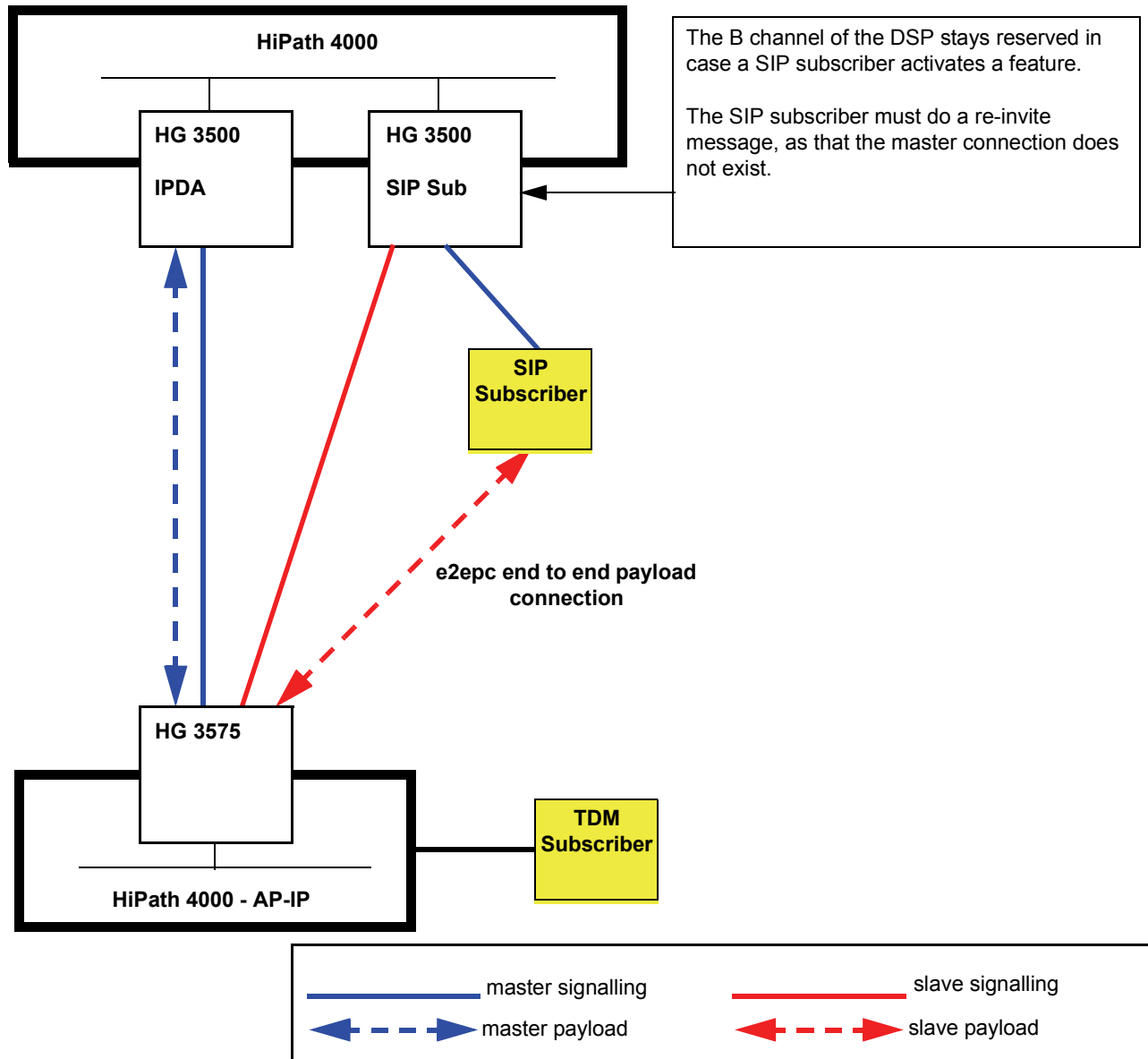
3.3.7 HFA Subscriber at HiPath 3000

Master signalling via HG 1500, master payload between HFA subscriber and neighboring HG 3500, slave signalling between HFA subscriber and initiating HG 3500 and slave payload between SIP and HFA subscriber.



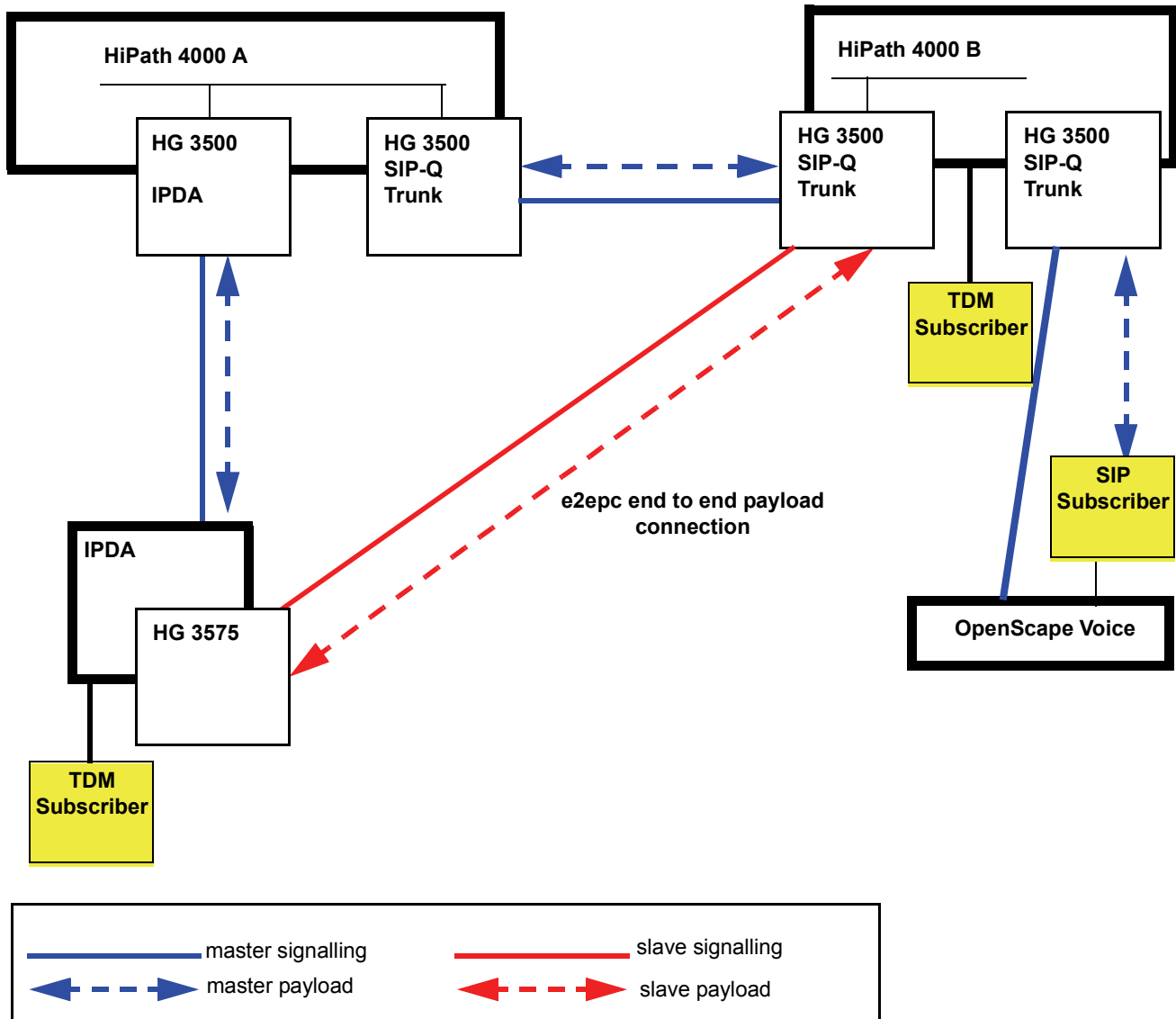
3.3.8 TDM Subscriber/Network behind a Remote Shelf of a HiPath 4000

DMC signalling between HG 3500 and HG 3575, e2epc between SIP subscriber and HG 3575.



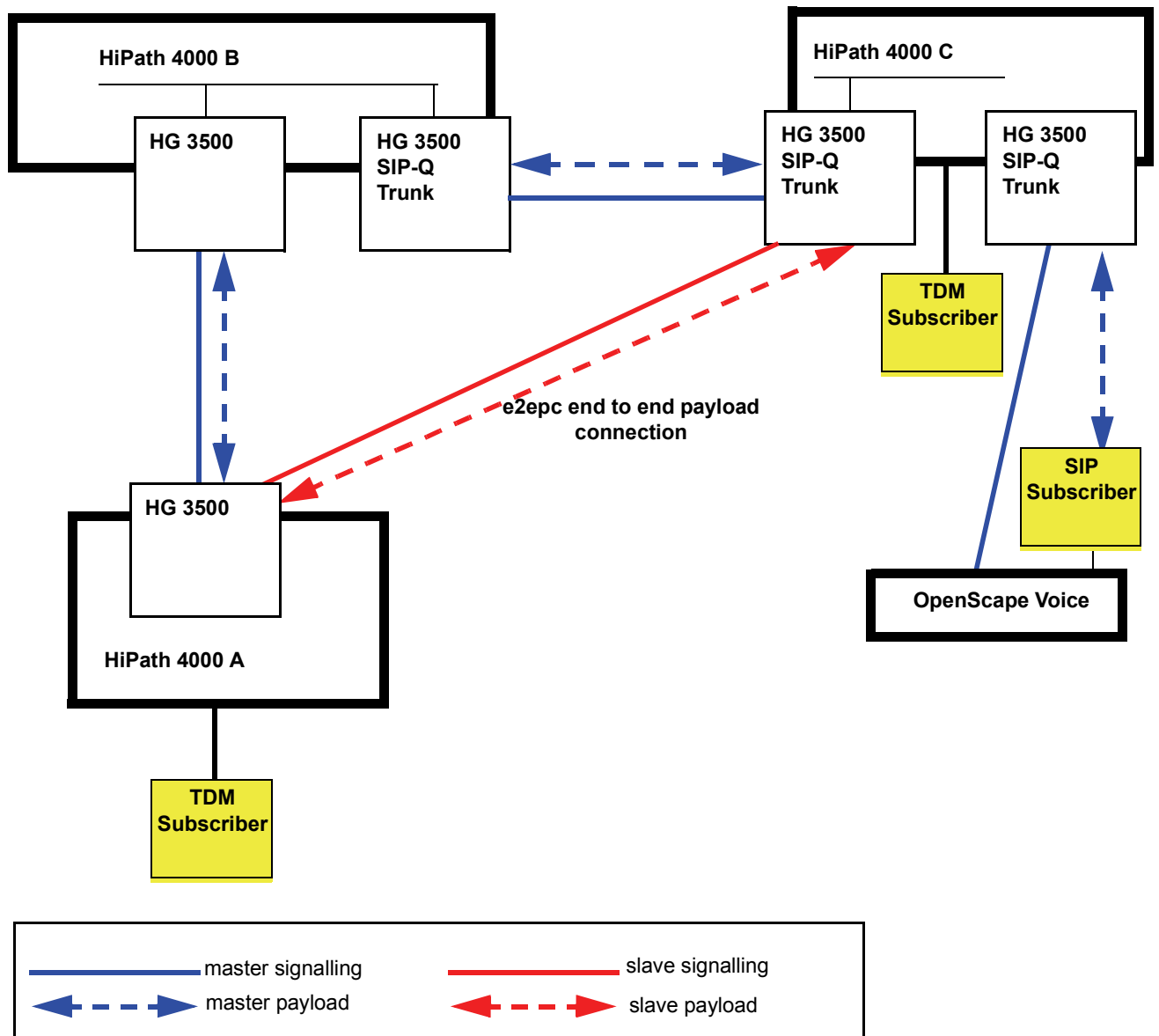
3.3.9 TDM Subscriber/Network at another HiPath 4000 behind a Remote Shelf

Slave signalling and payload to HG 3575.



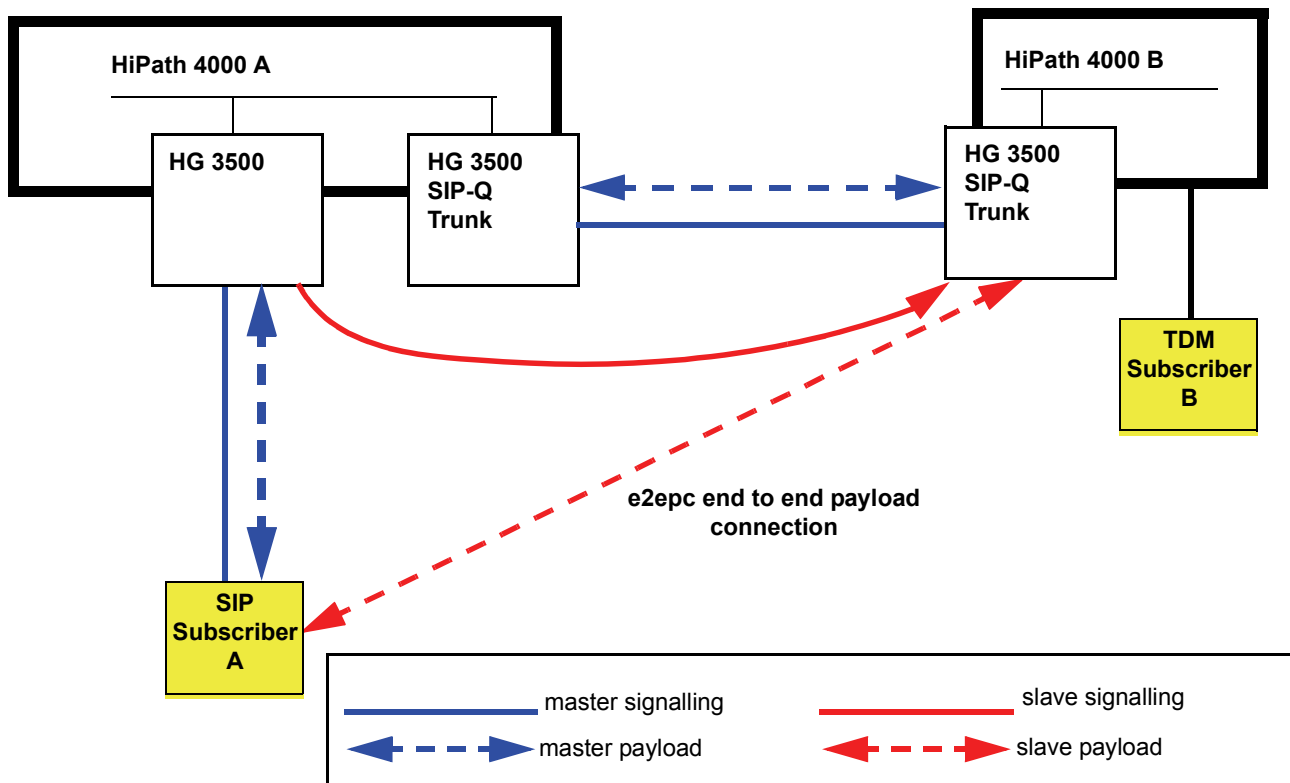
3.3.10 TDM Subscriber/Network at another HiPath 4000

Example 1: Slave signalling and payload to HG 3500.



Example 2: DMC signalling between HG 3500 (SIP subscriber) and HG 3500 (Trunk), e2epc between SIP subscriber and HG 3500.

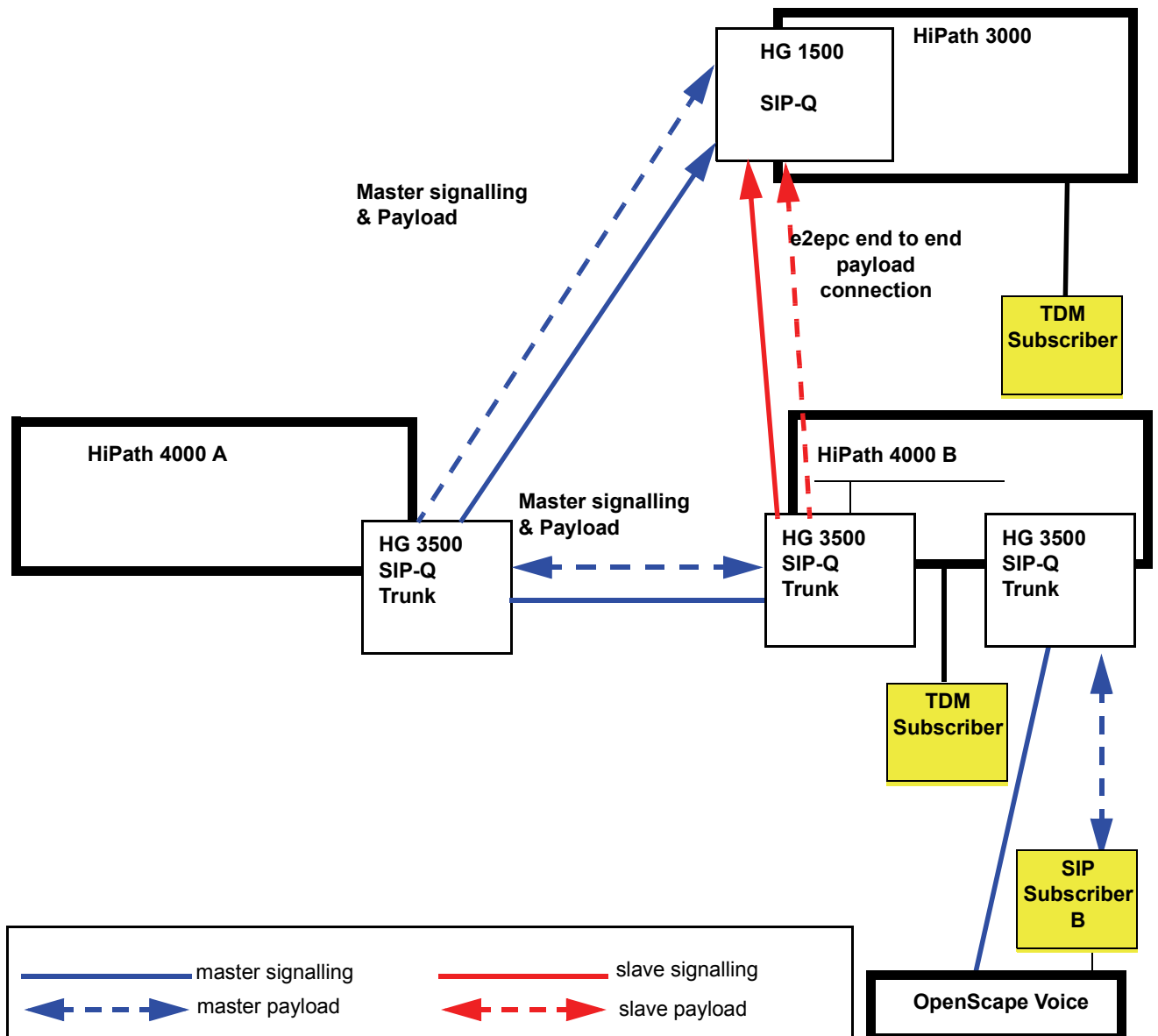
SIP subscriber A sets up a call to HiPath 4000 A. HiPath 4000 A routes the call via a trunking connection to HiPath 4000 B. HiPath 4000 B then sets up a connection to TDM subscriber B. HiPath 4000 B initiates a DMC between HG 3500 in HiPath 4000 B and the HG 3500 in HiPath 4000 A. HiPath 4000 A ensures that an end-to-end payload connection (e2epc) is established between SIP subscriber A and the HG 3500 in HiPath 4000 B.



3.3.11 TDM Subscriber at HiPath 3000

Master signalling and payload between HG 1500 and neighboring HG 3500, slave signalling and payload between HG 1500 and initiating HG 3500.

TDM subscriber at HiPath 3000 sets up a connection to HiPath 4000 A. HiPath 4000 A routes the call via a trunking connection to HiPath 4000 B. HiPath 4000 B routes the call via another trunking connection to OpenScope Voice. OpenScope Voice then sets up a connection to SIP subscriber B. HiPath 4000 B sets up a DMC between HG 3500 in HiPath 4000 B and HG 1500 in HiPath 3000.



3.3.12 Scenarios with HiPath 4000 SoftGate

Please refer to section „HiPath 4000 SoftGate“, [Chapter 8, “Direct Media Connect \(DMC\)”](#).

4 SIP Trunk Profiles

From HiPath 4000 V5 or later you are recommended to configure **native** SIP trunking connections (e.g. to a SIP service provider) via the WBM and to use SIP trunk profiles (see [Abschnitt 4.1, “Advantages”](#)). There are already SIP trunk profiles for SIP trunking connections available in the WBM (see [Abschnitt 4.2, “Default Profiles”](#)).

WICHTIG: From HiPath 4000 V5 or later native SIP trunk configuration is taken over by the gateway from the AMO CGWB, i.e. with an upgrade, the **Use Profiles** checkbox is automatically activated in the WBM (**WBM > Explorer > Voice Gateway > SIP Trunk Profile Parameters > Checkbox Use Profiles for Trunks via Native SIP**).

This means that following an update to HiPath 4000 V5, any native SIP trunks already configured or incorrectly configured (as native SIP) SIP-Q trunking connections no longer work. The parameters configured in the AMO CGWB (**Registrar, Proxy, GWRNR**) or AMO GKREG / AMO LDAT (destination IP) must be activated as profiles in the WBM. If you do not want this, the **Use Profiles** checkbox must be deactivated. **This is not recommended!**

4.1 Advantages

These are the advantages of using SIP trunk profiles:

- Increased security
If (standard)[Proxy](#), [Inbound proxy](#) and/or [Outbound proxy](#) were configured via the WBM, with each incoming call a check is carried out as to whether this originates from a valid IP address. If not, the call is rejected.
- DNS SRV support
DNS SRV can be used to propagate via DNS which IP-based services (e.g. SIP) in a domain (e.g. SIP provider domain) are provided.
- SIP specifications support of the trunk partner
The parameters of the trunk profile describe the SIP specifications (e.g. SIP header) of the trunk partner (e.g. SIP provider).

4.2 Default Profiles

- Five 5 SIP trunk profiles for native SIP trunking scenarios are already provided with default values:

SIP Trunk Profiles

Activating the SIP trunk profiles

- Arcor
- Broadsoft
- MediatrixGateway
- NatTrkWithoutRegistration (native SIP trunking without registration)
- NatTrkWithoutRegistration (native SIP trunking with registration)

The default profiles contain specific provider profiles which have been tested and released. The list of the default profiles can be extended with later minor releases / hotfixes.

Proxy

Server name / IP address of the server of the partner to which calls are routed.

Inbound proxy

Server name / IP address of the trunking partner from which the data are sent to the HiPath.

Outbound proxy

Server name/IP address to which all outgoing messages/data are sent as first node/hop (e.g. session boarder controller).

4.3 Activating the SIP trunk profiles

By default the SIP trunk profiles are activated in the WBM. If this feature was deactivated, it must be activated in the respective board / HiPath 4000 SoftGate in the WBM.

Call up:

Activate **WBM > Explorer > Voice Gateway > SIP Trunk Profile Parameter**
> (right-click) Checkbox **"Use Profiles for Trunks via Native SIP"**



Bild 4 Activating SIP trunk profiles

The menu **SIP Trunk Profiles** is then visible in the WBM under voice gateway.

4.4 "SIP Trunk Profiles" menu

HINWEIS: In the **SIP Trunk Profile Parameter** menu the **Use Profiles** checkbox must be activated. Otherwise the **SIP Trunk Profiles** menu is not displayed (see [Abschnitt 4.3, "Activating the SIP trunk profiles"](#)).

Call up:

WBM > Explorer > Voice Gateway > SIP Trunk Profiles

Five 5 SIP trunk profiles for native SIP trunking scenarios are already provided with default values (see [Abschnitt 4.2, "Default Profiles"](#)).

SIP-Trunk-Profil

Provider-Name: Arcor
 Account/Authentifizierung nötig: Nein
 Domain-Name: arcor.de

Registrar
 Registrar verwenden: Nein
 IP Adresse/Host-Name:
 Port: 0
 Reregistration-Intervall (s): 0

Proxy
 IP Adresse/Host-Name: 212.144.52.16
 Port: 0

Outbound-Proxy
 Outbound-Proxy verwenden: Ja
 IP Adresse/Host-Name: 218.1.31.140
 Port: 0

Inbound-Proxy
 Inbound-Proxy verwenden:
 IP Adresse/Host-Name:
 Port:

CLIP / CLIR
 CLIP outgoing in From header - display part: call number
 CLIP outgoing in From header - user part: call number
 CLIP outgoing in P-Asserted-Id header - display part: call number
 CLIP outgoing in P-Asserted-Id header - user part: call number
 CLIP outgoing in P-Preferred-Id header - display part: omit
 CLIP outgoing in P-Preferred-Id header - user part: omit
 CLIR outgoing in From header - display part: omit
 CLIR outgoing in From header - user part: call number
 CLIR outgoing Privacy header: id

Call number formatting
 Incoming call - Called party number: request line
 Incoming call - Calling party number: automatic
 Incoming call - Type of number (calling): automatic
 Outgoing call - Type of number (calling): automatic
 Incoming call - Type of number (called): automatic
 Outgoing call - Type of number (called): automatic

Miscellaneous
 Use route URI authentication: Ja
 Hold mode: send only
 Ignore 100 Rel: Nein
 Host Part of Contact header: IP Address
 ContactUriWithProtocol: Ja
 DirectPayload: Ja
 UseViaRPort: Ja
 RedirNrnFrom: Ja

Bild 5

Example of a standard profile:

4.4.1 Change

If the profiles already defined do not fit exactly, this can be changed.

SIP Trunk Profiles

"SIP Trunk Profiles" menu

The following connection parameters can be modified.

- **Provider name** (,
- **Account/authentication required** (see [Abschnitt 4.4.3, "Authentication"](#))
- **Use registrar**
- **Proxy**
- **Outbound proxy**
- **Inbound proxy**

The features available in the sections **CLIP/CLIR**, **Call number formatting** and **Miscellaneous** cannot be changed. If anything needs to be adapted here, a new profile must be created.

4.4.2 Activate

If the profile has been configured, it must then be activated.

Procedure:

1. Marking a SIP trunk profile.
2. Select **Activate** from the menu (right-click).

If the profile was activated successfully, the entire profile turns green.



Bild 6

Activated SIP trunk profile (Example)

4.4.3 Authentication

If the partner requires the "Authentication" feature to be active, this must be configured for the corresponding profile. The corresponding data (login and password) are specified by the trunking partner. Additionally the trunking connection with registration must then naturally work.

Procedure:

1. In the SIP trunk profile menu activate the **Account/Authentication required** checkbox.
2. Right-click **Add Account/Authentication** from the menu and enter the data provided by the trunking partner).



Bild 7

Add Account/Authentication (Example)

3. Activate Users

4.4.4 Deactivate

Procedure:

1. Marking a SIP trunk profile.
2. Select (right-click) **Deactivate** from the menu.

If the profile is successfully deactivated, the green coloring is removed.

4.4.5 Deletion

Before a SIP trunk profile can be deleted, all accounts must first be deleted. Only then can the SIP trunk profile be deleted.

Procedure:

1. Marking a SIP trunk profile.
2. Select (right-click) Delete from the menu.

The selected profile is deleted.

SIP Trunk Profiles

"SIP Trunk Profiles" menu

5 SIP-Q Trunking

5.1 Networking with SIP-Q-Trunking

SIP-Q-Trunking can be used in the following networking scenarios:

- HiPath 4000 - HiPath 4000
- HiPath 4000 - HiPath 2000
- HiPath 4000 - HiPath 3000/5000
- HiPath 4000 - OpenScape Voice

5.2 Configuration

see [Chapter 7, “SIP-Q-Trunking / Native SIP Trunking Configuration”](#).

6 Native SIP Trunking

Native SIP trunks can be used to connect to any external system, SIP service provider (see [Chapter 8, “SIP Service Provider”](#)) or 3rd party SIP products (e.g. OpenScape Xpressions).

6.1 Features

The following features are supported via native SIP trunks:

- Basic call
- Hold/retrieve
- Display Name Identification Services (DSS1) (available on the subscriber interface and on the native SIP trunking interface (Cornet-NQ))
- CLIP, CLIR, COLP, COLR
- Call forwarding
- Conferencing (3 party conference)
- Call transfer (**attended transfer** and **blind transfer**)
- Single step transfer
- Message waiting indication
- T.38 Fax

Additional features are not supported. HiPath systems are normally networked via SIP-Q (see [Chapter 5, “SIP-Q Trunking”](#)).

attended transfer

A is the initiator of the call transfer. A is in a call with B and puts B on hold. Then A sets up a consultation call to C and initiates the transfer. After the transfer, B and C are connected.

blind transfer

A is the initiator of the call transfer. A is in a call with B. A puts B on hold. Then A sets up a consultation call to C and transfers the call before the consultation call is connected. After the transfer, B and C are connected.

6.2 Native SIP Trunking Partners

- Service Provider:

- OpenScape Xpressions

Call Transfer

If a user wants to be transferred by the Xpressions the transfer is done using REFER with or without Replaces header field.

6.3 Configuration

6.3.1 General Configuration

Please refer to [Chapter 7, “SIP-Q-Trunking / Native SIP Trunking Configuration”](#) or [Chapter 8, “SIP Service Provider”](#).

6.3.2 Codec Configuration

OpenScape Xpressions supports the following codecs:

- G711-A,
- G711-μ
- G729AB.

Codec configuration is performed via CHANGE-CGWB or via WBM:

```
CHANGE-  
CGWB:MTYPE=CGW,LTU=<LTU>,SLOT=<SLOT>,TYPE=ASC,PRIO=PRIO1,CODE  
C=<CODEC>,RTP=<RTP>;
```

7 SIP-Q-Trunking / Native SIP Trunking Configuration

7.1 Configuration using AMOs

IMPORTANT: As of HiPath 4000 V5 it is recommended to configure native SIP trunks with the SIP trunk profiles function in the Web Based Management. For more details please refer to [Chapter 4, “SIP Trunk Profiles”](#) or [Section 7.2, “Configuration using SIP Trunk Profiles in the WBM”](#)

7.1.1 Trunking Protocol

The same protocols must always be set in the AMO CGWB and AMO GKREG.

AMO CGWB	AMO GKREG	Protocol
TRPRSIP	SIP	native SIP
TPRSIPQ	SIPQ	SIP-Q V2

Table 4 Protocols in AMO CGWB and AMO GKREG

7.1.2 Configuring an HG 3500 as a Local Gateway

The following diagram shows a HG 3500 scenario for a fictitious customer in Frankfurt. It shows node 10-69-100 (Frankfurt-Bockenheim) and node 10-69-200 (Frankfurt-Sachsenhausen). Both systems have a common gateway board that should be configured as HG 3500. LEGK should be active on both systems (two local gateways). This constellation is relatively easy to explain because both nodes can be configured more or less symmetrically. The figure shows all necessary parameters. Node 10-69-100 is located in IP network 1.69.11.0/24 and node 10-69-200 is located in IP network 1.69.21.0/24. The networks reach each other over the routers R_a and R_b .

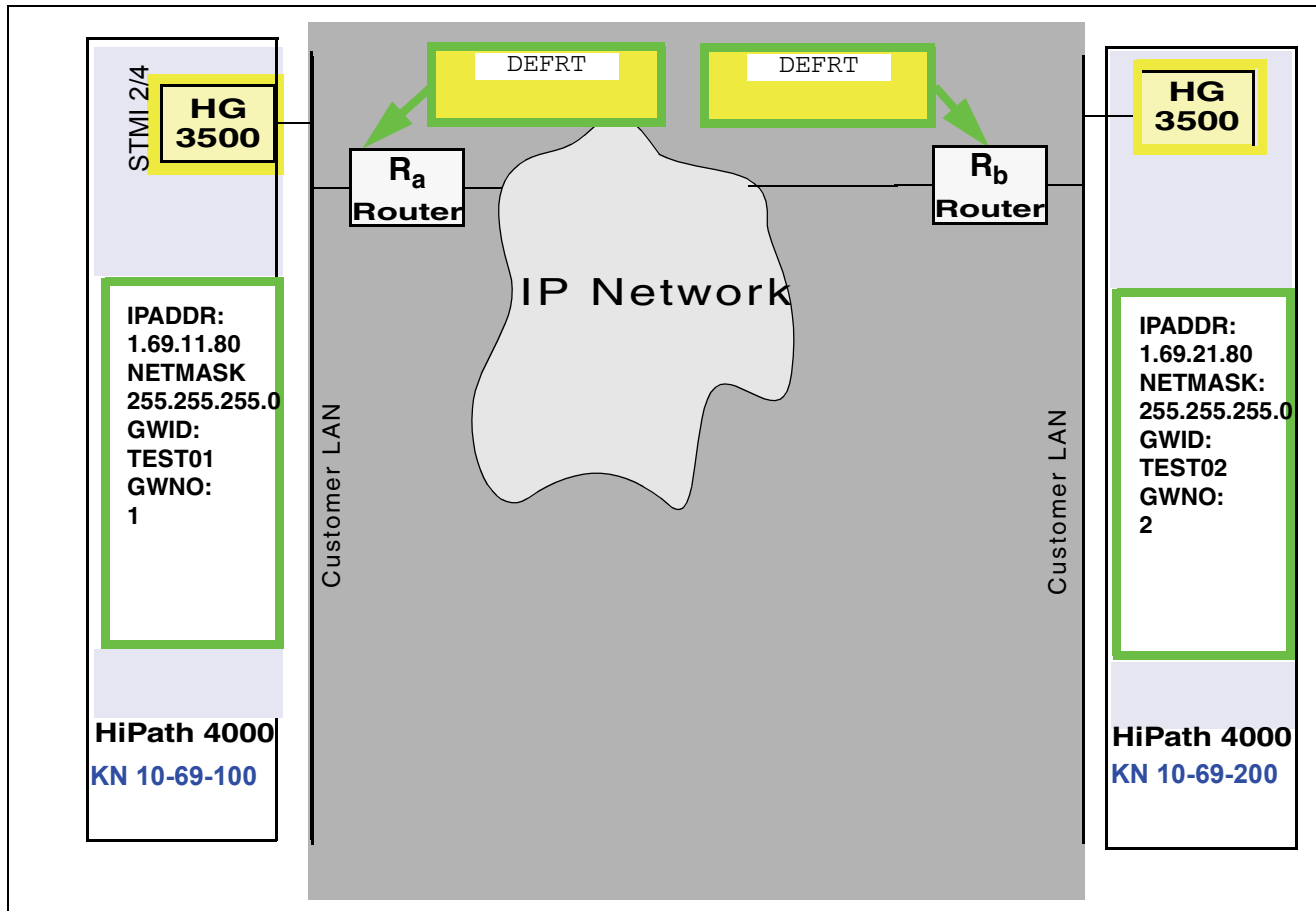


Figure 8 Configuration example: both nodes are LEGK

7.1.2.1 Configuration in Node 10-69-100

Start by configuring the board as HG 3500 in node 100. To avoid repeated restarts, do not insert the board until configuration is complete.

Step 0:

Configure LEGK in node 10-69-100.

```
CHANGE-ZANDE:TYPE=ALLDATA,GATEKPR=YES;
```

Step 1:

The Common Gateway Board Q2316-X is configured as HG 3500. Q2316-X is the version with 60 B channels. You can configure the board as a pure IP trunking board or in mixed operation with IP trunking and LAN connectivity (WAML) and many other features.

Configuration of the board for IP trunking only:

```
ADD-BFDAT:FCTBLK=1,FUNCTION=HG3550,BRDBCHL=BCHAN60&BCHAN120;
```

```
CHANGE-
BFDAT:CONFIG=CONT,FCTBLK=1,FUNCTION=HG3550,LINECNT=1,UNITS=3; /*
30 B channels for HG3550 functionality (H323 trunking, SIP
trunking, SIP subscriber)

CHANGE-BFDAT:CONFIG=OK,FCTBLK=1,ANSW=YES; /* Close the
configuration

ADD-BCSU:TYPE=IPGW,LTG=1,LTU=1,SLOT=37,PARTNO="Q2316-X
",FCTID=1,LWVAR="0",FCTBLK=1,BCHAN3550=30,ALARMNO=0;
```

or

IP Trunking & WAML:

```
ADD-
BFDAT:FCTBLK=1,FUNCTION=HG3550&WAML,BRDBCHL=BCHAN60&BCHAN120;

CHANGE-
BFDAT:CONFIG=CONT,FCTBLK=1,FUNCTION=HG3550,LINECNT=1,UNITS=3; /*
30 B channels for HG3550 functionality (IP trunking, SIP
trunking, SIP subscriber)

CHANGE-BFDAT:CONFIG=CONT,FCTBLK=1,FUNCTION=WAML,UNITS,LINECNT=1;
/* Board will be configured with one circuit, i.e. 10 B channels
for WAML

CHANGE-BFDAT:CONFIG=OK,FCTBLK=1,ANSW=YES; /* Close the
configuration

ADD-BCSU:TYPE=IPGW,LN=1,LTU=1,SLOT=37,PARTNO="Q2316-X
",FCTID=1,LWVAR="0",FCTBLK=1,BCHL3550=30,BCHLWAML=10,ALARMNO=0;
```

Step 2:

The AMO CGWB allows the board to receive the IP address in the customer LAN (LAN1), the subnet mask and the protocol variants (H323A / SIP-Q / NONE (board is only configured for SIP subscribers).

IMPORTANT: Only one trunking protocol may be configured for each board!
There are no restrictions on interworking with other functions!

```
ADD-
CGWB:LTU=1,SLOT=37,SMODE=NORMAL,IPADDR=1.69.11.80,NETMASK=255.25
5.255.0,TRPRSIPQ=10; /*10 B channels for SIP-Q trunking
```

or

```
ADD-
CGWB:LTU=1,SLOT=37,SMODE=NORMAL,IPADDR=1.69.11.80,NETMASK=255.25
5.255.0,TRPRSIP=10; /*10 B channels for native SIP trunking
```

=> 20 B channels remain for other functions

Step 3:

The default router is now set in the customer LAN:

```
CHANGE-
CGWB:MTYPE=CGW,LTU=1,SLOT=37,TYPE=GLOBIF,DEFRT=1.69.11.254;
```

Step 4:

The HG 3500 boards are administered using one gateway number (**GWNO**). This number must also be unique.

```
CHANGE-CGWB:MTYPE=CGW,LTU=1,SLOT=37,TYPE=LEGKDATA,GWNO=1;
```

Step 5:

Configuration without authentication and without registration (default values in AMO CGWB).

```
CHANGE-  
CGWB:MTYPE=CGW,LTU=1,SLOT=37,TYPE=LEGKDATA,GWNO=1,REGEXTGK=NO;
```

SSA:

```
CHANGE-CGWB:MTYPE=CGW,LTU=1,SLOT=37,TYPE=SIPTRSSA,SIPREG=NO;
```

ERH:

```
CHANGE-CGWB:MTYPE=CGW, . . . ,TYPE=SIPTRERH,GWAUTREQ=NO;
```

IMPORTANT: Parameter **SIPREG** and **REGEXTGK** must always have the same value.

Step 6:

IMPORTANT: Multiple boards are permitted in one trunk group!

This step consists of several configuration commands. We need to set up the trunk group and circuit, via which the B channels between gateway and HiPath 4000 are switched.

First you need a trunk group:

```
ADD-BUEND:TGRP=50,NAME="IP TRUNK GWID 1 ",NO=30;
```

A digital trunk is then required (AMO TDCSU). The device type is **HG3550IP** (for H323/H323A and native SIP/SIP-Q trunking). If you want to permit DMC, set **DMCALLWD=Y**. This is only possible when the partner system supports DMC. The <number> in the parameter **LWPAR** must identify the trunk as MASTER.

```
ADD-TDCSU:OPT=NEW,PEN=1-01-037-0,COTNO=36,COPNO=32,DPLN=0,ITR=0,  
COS=2,LCOSV=1,LCOSD=1,CCT="CIRCUIT from  
GW2",PROTVAR="ECMAV2",SEGMENT=8,  
ISDNIP=0,ISDNNP=0,TRACOUNT=31,SATCOUNT=MANY,NNO=1-69-  
1499,ALARMNO=12,COTX=36,CCHDL=SIDEA,CLASSMRK=EC&G711,TCCID="IP  
RS",TGRP=50,SRCHMODE=DSC,  
INS=Y,DEV=HG3550IP,BCHANNEL=1&&30,BCNEG=N,BCGR=1,LWPAR=0,DMCALLW  
D=YES;
```

If DMC has been activated for a native SIP/SIP-Q trunking path, Voice Activity Detection should also be activated. Otherwise the same bandwidth is required for the master connection as for the slave connection!

CHA-

CGWB:MTYP=CGW,LTU=<ltu>,SLOT=<slot>,TYP=ASC,PRIO=PRIO1,**VAD=YES**;

Otherwise the same bandwidth is required for the master connection as for the slave connection.

IMPORTANT: It is recommended to reset the board:

RESET-BSSU:ADDRTYPE=PEN,LTG=1,LTU=1,SLOT=37;

IMPORTANT: The command CHANGE-FUNSU can prevent the complete reloading of the STMI board.

Reloading is only necessary in the case of loadware changes.

CHANGE-FUNSU:PIT=FLASH,PARTNO="Q2316-X
",FCTID=2,ACTION=RESET;

Step 7:

The LEGK must now still be informed about the existence of the HG 3500 gateway in the AMO GKREG. The **GWNO** must match the entry from the AMO CGWB. In the case of a local gateway, the AMO GKREG automatically loads the IP address (**IPADR**) from the AMO CGWB. You must also set the attributes **INTGW** (internal -local- gateway) and **HG3550V2** (version ID). The parameter **REGGW** is not necessary because the gateway does not have to register at another gateway.

ADD-GKREG:**GWNO=1**,GWATTR=**INTGW&HG3550V2&SIPQ** or
SIP,DIPLNUM=0,DPLN=0,LAUTH=1;

If a local gateway is configured, **REGISTERED=NO** is always displayed with
DISPLAY-GKREG;.

Step 8:

The same transport protocol (TCP, UDP) must be configured in all systems. With WBM, the transport protocol used can be set under **Explorer > Voice Gateway > SIP Parameters**.

SIP-Q-Trunking / Native SIP Trunking Configuration

Configuration using AMOs

SIP Parameters

SIP User Agent

Use SIP Registrar: No
SIP Registrar IP Address: 0.0.0.0
SIP Registrar Port Number: 0
Period of registration (sec): 0

SIP Server (Registrar / Redirect)

SIP Server IP Address: 172.28.145.101
Period of registration (sec):

SIP Transport Protocol

SIP via TCP: Yes
SIP via UDP: ☒

Now the HG 3500 board in node 10-60-100 is configured properly and is communicating with the LEGK. However, the gateway is still missing in the partner node.

7.1.2.2 Configuration in Node 10-69-200

An LEGK should also be active in node 10-69-200. As the configuration procedures for nodes 10-69-200 and 10-69-100 are identical, this section only lists the appropriate AMO commands for the steps described above.

Step 0:

```
CHANGE-ZANDE:TYPE=ALLDATA,GATEKPR=YES;
```

Step 1:

```
ADD-BFDAT:FCTBLK=1,FUNCTION=HG3550,BRDBCHL=BCHAN60&BCHAN120;  
CHANGE-  
BFDAT:CONFIG=CONT,FCTBLK=1,FUNCTION=HG3550,LINECNT=1,UNITS=3;  
CHANGE-  
BFDAT:CONFIG=CONT,FCTBLK=1,FUNCTION=HG3550,LINECNT=1,UNITS=3; /*  
30 B channels for HG3550 functionality (IP trunking, SIP  
trunking, SIP subscriber)  
CHANGE-BFDAT:CONFIG=OK,FCTBLK=1,ANSW=YES; /* Close the  
configuration
```

```
ADD-BCSU:TYPE=IPGW,LN=1,LTU=2,SLOT=49,PARTNO="Q2316-X  
",FCTID=1,LWVAR="0",FCTBLK=1,BCHAN3550=30,ALARMNO=0;
```

Step 2:

```
ADD-  
CGWB:LTU=2,SLOT=49,SMODE=NORMAL,IPADDR=1.69.21.80,NETMASK=255.25  
5.255.0,TRPRSIPQ=10;
```

or

ADD-
CGWB:LTU=2, SLOT=49, SMODE=NORMAL, IPADDR=1.69.21.80, NETMASK=255.255.255.0, **TRPRSIP=10**;

Step 3:

CHANGE-
CGWB:MTYPE=CGW, LTU=2, SLOT=49, TYPE=GLOBIF, DEFRT=1.69.21.254;

Step 4:

CHANGE-CGWB:MTYPE=CGW, LTU=2, SLOT=49, TYPE=LEGKDATA, GWNO=2;

Step 5:

CHANGE-
CGWB:MTYPE=CGW, LTU=2, SLOT=49, TYPE=LEGKDATA, GWNO=2, **REGEXTGK=NO**;

SSA:

CHANGE-CGWB:MTYPE=CGW, LTU=2, SLOT=49, **TYPE=SIPTRSSA, SIPREG=NO**;

ERH:

CHANGE-CGWB:MTYPE=CGW, LTU=2, SLOT=49, **TYPE=SIPTRERH, GWAUTREQ=NO**;

IMPORTANT: Parameters **SIPREG** and **REGEXTGK** must always have the same value.

Step 6:

ADD-BUEND:TGRP=50, NAME="IP TRUNK GWID 2 ", NO=30;

ADD-TDCSU:OPT=NEW, PEN=1-02-049-0, COTNO=36, COPNO=32, DPLN=0, ITR=0, COS=2, LCOSV=1, LCOSD=1, CCT="CIRCUIT from GW2", PROTVAR="ECMAV2", SEGMENT=8, ISDNIP=00, ISDNPN=0, TRACOUNT=31, SATCOUNT=MANY, NNO=1-69-1499, ALARMNO=12, COTX=36, CCHDL=SIDEA, CLASSMRK=EC&G711, TCCID="IP RS", TGRP=50, **SRCHMODE=DSC**, INS=Y, **DEV=HG3550IP**, BCHANNEL=1&&30, BCNEG=N, BCGR=1, LWPAR=0 (=Master), **DMCALLWD=YES**;

CHA-
CGWB:MTYP=CGW, LTU=<ltu>, SLOT=<slot>, TYP=ASC, PRIO=PRIO1, **VAD=YES**;

Step 7:

The GKREG contains the configuration for **both** HG 3500 boards. Here, **GWNO=2** is now the local gateway in node 10-69-200 with the attributes **INTGW** and **HG3550V2** (for H323/H323A and native SIP/SIP-Q) while **GWNO=1** (the partner gateway in node 10-69-100) contains the attributes **EXTGW**, **HG3550V2** and **native SIP/SIP-Q** in this system. The IP address and the **GWDIRNO** must be specified for the external gateway because the associated AMO CGWB is configured in the partner system. The parameter **REGGW** is not necessary because the gateway does not have to register at another gateway.

ADD-GKREG: **GWNO=1**, GWATTR=**EXTGW**&HG3550V2&**SIPQ or SIP**, GWIPADDR=1.69.11.80, DIPLNUM=0, DPLN=0, LAUTH=1;

```
ADD-GKREG:GWNO=2,GWATTR=INTGW&HG3550V2&SIPQ or  
SIP,DIPLNUM=0,DPLN=0,LAUTH=1;
```

7.1.2.3 Extension in Node 10-69-100

The HG 3500 in node 10-69-200 must be retrofitted in node 10-69-100 for LEGK (AMO GKREG). For node 10-69-100, this HG 3500 (**GWNO=2**) is an external gateway (attributes **EXTGW&HG3550V2&SIPQ/SIP**) and is not registered in node 10-69-100.

```
ADD-GKREG:GWNO=2,GWATTR=EXTGW&HG3550V2&SIPQ or  
SIP,GWIPADDR=1.69.21.80, DIPLNUM=0,DPLN=0,LAUTH=1;
```

Node 10-69-100

Now DIS-GKREG; should display in node 10-69-100:

- **GWNO=1** is registered and
- **GWNO=2** is not registered.

Node 10-69-200

Now DIS-GKREG; should display in node 10-69-200:

- **GWNO=1** is not registered and
- **GWNO=2** is registered.

7.1.2.4 Call from Node 10-69-100 to Node 10-69-200

LCR should now be configured for node 10-69-100 in our example so that a station using open numbering (**UNKNOWN**) in this node can reach a station in node 10-69-200 over the IP trunk. Closed numbering or ISDN numbering plans are also supported and configured in the same way.

A station in node 10-69-100 reaches node 10-69-200 over the tie number 902.

```
ADD-WABE:CD=902,DAR=TIE;
```

LRTE 520 should route to node 10-69-200 and use trunk group 50 (see above):

```
ADD-RICHT:MODE=LRTENEW,LRTE=520,LSVC=ALL,NAME="IP TO KN 69  
200",TGRP=50,DNNO=1-69-499;
```

The following is a simple outdial rule:

```
ADD-LODR:ODR=520,CMD=ECHO,FIELD=2;  
ADD-LODR:ODR=520,CMD=END;  
ADD-LODR:ODR=520,INFO="TIE TO GW2";
```

Enter **GW1=2-0** in LRTE 520 now. The **2** refers to the parameter **GWNO** in AMO GKREG (section [Section 7.1.2.3, "Extension in Node 10-69-100"](#)). In AMO GKREG for node 10-69-100, the HG 3500 with **GWNO=2** is the IP address

1.69.21.80. The **0** stands for the sector path number. Sector path 0 means there is unlimited bandwidth for this path. If the destination gateway is not reachable, you can use the parameters **GW2** to **GW5** to configure an alternative route.

ADD-

LDAT:LROUTE=520,LSVC=ALL,LVAL=1,TGRP=50,ODR=520,LAUTH=1,GW1=2-0;

IMPORTANT: In contrast to S0/S2 connections (point-to-point), an IP connection is a point-to-multipoint connection and therefore the same trunk number to the different destination gateways (**GW1-GW5**) in the AMO LDAT!

The tie number is entered in the AMO LDPLN. The directory number is entered in the default dial plan with **DIPLNUM=0**. A profile index can also be used instead of LRTE (AMO-LPROF):

ADD-LDPLN:LCRCONF=LCRPATT,DIPLNUM=0,LDP="902"-
"X",LROUTE=520,LAUTH=1;

If you want to be able to reach node 10-69-100 from node 10-69-200, perform the following configuration. You then point to the **GWNO=1** in AMO LDAT.

7.1.3 Backup & Restore of the WBM Configuration Data

All configuration data configured via WBM must be saved on a backup server. Otherwise all configuration data will be lost after a board replacement, a reset of the board or loadware upgrade and the native SIP/SIP-Q trunking connection can't be automatically reestablished. This is done with the HiPath Backup & Restore.

Therefore it is important to have a working connection to the backup server. If the backup server is not available, the configuration data can also be saved in the flash memory of the board and then can be retrieved from there.

HiPath Backup & Restore

Configuration of a connection to backup server and HiPath 4000 Assistant.

CHANGE-

CGWB:MTYPE=CGW,LTU=1,SLOT=37,TYPE=MGNTDATA,MGNTIP=<ip_address_assistant>,MGNTPN=<port_number_assistant>,BUSIP=<ip_address_backup_server>,BUSPN=<port_number_backup_server>;

Note: The values for **MGNTPN** und **BUSPN** won't be restored after a reset of the board. They have to be configured again manually.

Local Flash

IMPORTANT: It is recommended to additionally use the local flash as a backup media. If the connection to the backup server is broken the configuration data will be automatically restored from the local flash.

WBM > Maintenance > Actions > (double-click) Automatic Actions > Saving Local Configuration for Upgrade

For more information please refer to „HiPath Gateways HG 3500 and HG 3575“, Chapter 8, “Save Configuration Data in Local Flash”.

7.1.4 Displaying Link Status

The link status of the LAN interface (link signal Ethernet, LAN speed, and LAN interface) can be displayed at each level with the AMO SDSU using the usual commands (**PER1**, **PER2**, and **PER3**):

```
DISPLAY-SDSU:TYPE=PEN,LEVEL=PER1,LTG=1,LTU=<ltu no>,SLOT=<slot no>;
```

7.1.5 Procedure for Adding Supplementary HG 3500 Boards to the Network

Additional LEGKs can be configured in the same way as node 10-69-100. The AMO BFDAT, AMO BCSU, AMO CGWB, AMO GKREG and AMO LDAT are in a single node.

IMPORTANT: The GKREG must recognize **all** HG 3500 boards in the network.

7.1.6 Relevant AMOs

AMO	Parameter	Sprache/ Language	Beschreibung/ Description
BCSU	TYP=IPGW	d	CGW einrichten
	TYPE=IPGW	e	Configure CGW
	BKAN3550	d	Anzahl der B-Kanäle für IP-Trunking
	BCHAN3550	e	Number of b channels for IP trunking

AMO	Parameter	Sprache/ Language	Beschreibung/ Description
BFDAT	FUNCTION=HG3550	d	CGW mit IP-Trunking
	FUNCTION=HG3550	e	CGW with IP trunking
	BGBKAN	d	Maximale Anzahl der verfügbaren B-Kanäle der Baugruppe (entweder 60 oder 120)
	BRDBCHL	e	Maximum number of available b channels (60 or 120)
	ANZSATZ	d	Anzahl der Sätze für IP-trunking
	LINECNT	e	Number of lines for IP trunking
	ANZEINH	d	Anzahl der vordefinierten Blöcke mit ausgewählter Funktion pro Satz.
	UNITS	e	Defines the number of predefined blocks with the selected function per line.
CGWB	DEFRT	d	IP-Adresse des Default Routers
	DEFRT	e	IP address of the default router
	GWAUTREQ	d	Ziffern-Authentifizierung wird benötigt
	GWAUTREQ	e	Digest authentication is required
	GWREALM	d	Security Zone of the gateway
	GWREALM	e	Security Zone of the gateway
	GWRNR	d	Gateway Rufnummer
	GWDIRNO	e	Gateway Directory Number
	GWSECRET	d	Passwort des Gateways
	GWSECRET	e	Password of the gateway
	GWUSERID	d	User Identifizierung des Gateways
	GWUSERID	e	User identification of the gateway
	REGIP1	d	IP-Adresse des primären SIP Registrars
	REGIP1	e	IP address of the primary SIP registrar
	REGIP2	d	IP-Adresse des sekundären SIP Registrars
	REGIP2	e	IP address of the secondary SIP registrar
	PORTTCP1	d	Port Nummer für die primäre SIP Registrierung über TCP/UDP
	PORTTCP1	e	Port number of the primary SIP registration via TCP/UDP
	PORTTCP2	d	Port Nummer für die sekundäre SIP Registrierung über TCP/UDP
	PORTTCP2	e	Port number of the secondary SIP registration via TCP/UDP
	REGTIME	d	Zeit in Sekunden wie lange die SIP-Registrierung gültig ist

SIP-Q-Trunking / Native SIP Trunking Configuration

Configuration using SIP Trunk Profiles in the WBM

AMO	Parameter	Sprache/ Language	Beschreibung/ Description
	REGTIME	e	Time in seconds the SIP registration is valid
	SIPREG	d	Gateway an einem SIP Registrar registrieren
	SIPREG	e	Register gateway at a SIP registrar
	TRPRSIP	d	Anzahl der B-Kanäle für Trunking-Protokoll SIP
	TRPRSIP	e	Number of b channels for trunking protocol SIP
	TRPRSIPQ	d	Anzahl der B-Kanäle für Trunking-Protokoll SIP-Q
	TRPRSIPQ	e	Number of b channels for trunking protocol SIP-Q
	TYP=SIPTRSSA	d	SIP Trunking Daten für SSA (SIP Stack Agent)
	TYPE=SIPTRSSA	e	SIP trunking data for SSA (SIP Stack Agent)
	TYP=SIPTRERH	d	SIP Trunking Daten für ERH (Endpoint Registration Handler)
	TYPE=SIPTRERH	e	SIP trunking data for ERH (Endpoint Registration Handler)
	VAD	d	Voice Activity Detection
	VAD	e	Voice Activity Detection
GKREG	GWATTR	d	Gateway Attribut
	GWATTR	e	Gateway attribute
	GWNR	d	Gateway ID (muss dem Eintrag im AMO CGWB entsprechen)
	GWNO	e	Gateway ID (must correspond to the value in AMO CGWB)
TDCSU	DMCERL=NEIN	d	DMC wird verhindert
	DMCALLWD=NO	e	DMC not allowed
ZANDE	GATEKPR=JA	d	PBX mit Gatekeeper
	GATEKPR=YES	e	PBX with gatekeeper

7.2 Configuration using SIP Trunk Profiles in the WBM

IMPORTANT: This section is only valid for the configuration of SIP trunking connections!

7.2.1 Configuration without Registration

1. Verify that the SIP protocol variant for IP networking is set to native SIP in AMO CGWB (parameter **TRPRSIP**) and **Use Profiles for Trunks via Native SIP** flag is set (profile usage is activated) in the WBM.

Explorers > Voice Gateway > SIP Trunk Profiles > (right mouse) Activate check box **Use Profiles for Trunks via Native SIP**

SIP Trunk Profile Parameter

Signaling Protocol for IP Networking:	SIP
SIP Protocol Variant for IP Networking:	Native SIP
Use Profiles for Trunks via SIP-Q:	<input type="checkbox"/>
Use Profiles for Trunks via Native SIP:	<input checked="" type="checkbox"/>

Figure 9 Activate native SIP trunk profiles

2. Das angebotene Profil "NatTrkWithoutRegistration" kann für Provider-Konfigurationen ohne Registrierung verwendet werden. Die Provider **IP-Adresse/Host-Name** (z.B. IP 1.2.3.4) muss im Abschnitt **Proxy** konfiguriert werden.

Explorer > Sprachgateway > SIP-Trunk-Profil > NatTrkWithoutRegistration > (rechte Maustaste) **Ändern**

3. The offered **NatTrkWithoutRegistration** profile can be used for provider configuration without registration. The provider **IP Address/Host Name** (e.g. IP=1.2.3.4) has to be configured in the section **Proxy**.

Explorers > Voice Gateway > SIP Trunk Profiles > NatTrkWithoutRegistration > (right mouse) **Change**

SIP Trunk Profile

Provider Name:	NatTrkWithoutRegistration
Account/Authentication Required:	No
Domain Name:	

Registrar
Use Registrar: No
IP Address / Host name:
Port: 0
Reregistration Interval (sec) 0

Proxy
IP Address / Host name: 1.2.3.4
Port: 5060

Outbound Proxy
Use Outbound Proxy: No
IP Address / Host name: 0.0.0.0
Port: 0

Figure 10 IP address / host name of the provider (without registration)

4. Now the profile has to be activated. If the activation is successful the whole profile is shown in green.

Explorers > Voice Gateway > SIP Trunk Profiles > NatTrkWithoutRegistration > (right mouse) Activate

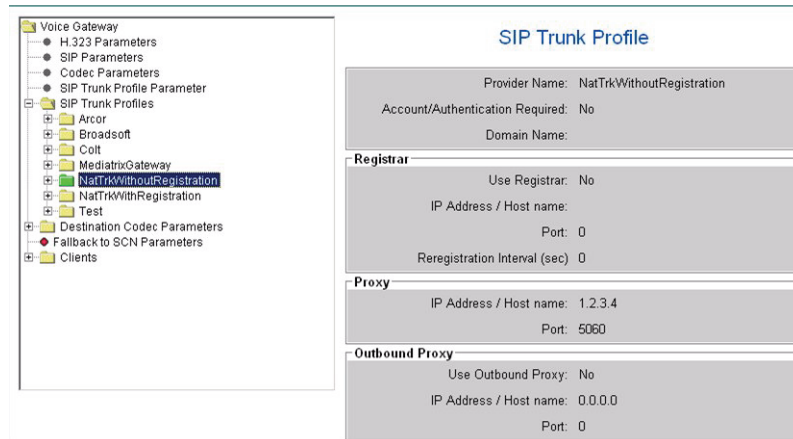
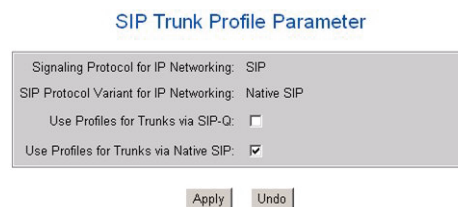


Figure 11 Activated SIP trunk profile (without registration)

7.2.2 Configuration with Registration

1. Verify that the SIP protocol variant for IP networking is set to native SIP in AMO CGWB (parameter **TRPRSIP**) and **Use Profiles for Trunks via Native SIP** flag is set (profile usage is activated) in the WBM.

Explorers > Voice Gateway > SIP Trunk Profiles > (right mouse) Activate
check box **Use Profiles for Trunks via Native SIP**



2. The offered **NatTrkWithRegistration** profile can be used for provider configuration with registration. The account used for registration and if required authentication data are configured as shown in the following screen shot.

Explorers > Voice Gateway > SIP Trunk Profiles > NatTrkWithRegistration > (right mouse) Change > Activate check box **Account/Authentication required**

Account Name

Account Name:	<input type="text" value="12345"/>
Authorization name:	<input type="text" value="UserAuth"/>
New Password:	<input type="password" value="*****"/>
Confirm Password:	<input type="password" value="*****"/>

Figure 12 Account/Authentication data

- The provider **IP Address/Host Name** (e.g. DNS name) has to be configured in the section **Registrar** and in the section **Proxy**. The **Domain Name** has normally the same value as **IP Address/Host Name** (see sections **Registrar** or **Proxy**) and is used as host part of URI in SIP messages.

Explorers > Voice Gateway > SIP Trunk Profiles > NatTrkWithRegistration > (right mouse) Change

SIP Trunk Profile

Provider Name:	NatTrkWithRegistration
Account/Authentication Required:	Yes
Domain Name:	provider.com

Registrar
Use Registrar: Yes
IP Address / Host name: provider.com
Port: 0
Reregistration Interval (sec) 1200

Proxy
IP Address / Host name: provider.com
Port: 0

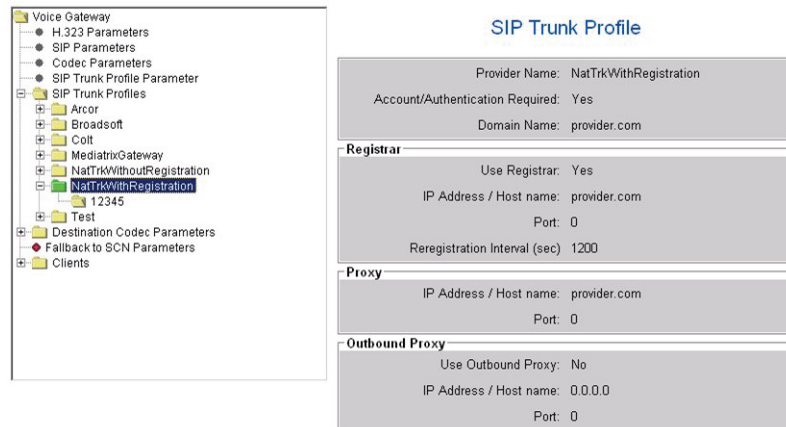
Outbound Proxy
Use Outbound Proxy: No
IP Address / Host name: 0.0.0.0
Port: 0

- Now the profile has to be activated. If the activation is successful the whole profile is shown in green.

Explorers > Voice Gateway > SIP Trunk Profiles > NatTrkWithRegistration > (right mouse) Activate

SIP-Q-Trunking / Native SIP Trunking Configuration

Configuration using HiPath 4000 Assistant



7.2.3 Notes

- Only one profile can be activated.
- Outbound Proxy has to be configured when Session Border Controller SBC (e.g. Comdasys box (LAN IP)) is used for LAN/WAN Nat traversal.

7.3 Configuration using HiPath 4000 Assistant

7.3.1 Configure Common Gateway HG 3500

In order to configure CGW board do the following:

- [Step 1: Configure function block](#)
- [Step 2: Search for a free slot](#)
- [Step 3: Open Board configuration menu in HiPath 4000 CM Assistant](#)
- [Step 4: Add new board](#)
- [Step 5: Filling in the mask](#)
- [Step 6: Open STMI2-IGW Board Data dialog and fill mandatory fields](#)

Step 1: Configure function block

Start **CM > System Data > Board > CGW Function Block**

Click **Search** for searching for a free function block number. Click **New** for adding a new function block. Enter the required data and click **Save**.

CGW Function Block

Object Edit View Action Scheduled Batch Extras

View: ☐ Search Criteria ☒ New Object

Function block: ☒ Finish configuration of function block

Dedicates the block for boards with

☒ 60 b-channels ☒ 120 b-channels

Function	Number of lines	Number of predefined blocks	Number of b-channels
<input checked="" type="checkbox"/> HG3530			
<input checked="" type="checkbox"/> HG3550	100		
<input checked="" type="checkbox"/> HG3570			
<input checked="" type="checkbox"/> WAML			
<input checked="" type="checkbox"/> SIP			
<input type="checkbox"/> HG3530R			
<input type="checkbox"/> HG3550R			
<input type="checkbox"/> SIPR			
<input type="checkbox"/> STANDBY			

Step 2: Search for a free slot

Before entering LTU and SLOT number it is recommended to check if desired SLOT is free by **CM > System Data > Maintenance > Board Maintenance**.

After search is performed in Object list view free LTU and SLOT is marked with **NOGEN/NPR/UNACH** under **Status Overview** column.

...	S...	System	Domain	Board...	Part Nu...	Status Overview	RE...	NPR	VBT	SL...	LO...	DE...	N
1	1	103	SYS1	POMAIN	STM20A	Q2316-X1	WBT/UNACH						
1	1	49	SYS1	POMAIN	STM20G	Q2316-X1	NPR/UNACH						
1	1	55	SYS1	POMAIN	STM20W	Q2316-X1	WBT/UNACH						
1	1	61	SYS1	POMAIN	SLM016	Q2164-X	NPR						
1	1	67	SYS1	POMAIN	SLM024	Q2168-X	READY						
1	1	73	SYS1	POMAIN	SLMG	Q2153-X1	NPR						
1	1	79	SYS1	POMAIN	SLMB	Q2150-X	WBT						
1	1	85	SYS1	POMAIN	SLMAR	Q2480-X	NPR						
1	1	91	SYS1	POMAIN	SLMA24	Q2246-X	NPR						
1	1	97	SYS1	POMAIN	TMED	Q2025-X3	WBT						
1	2	103	SYS1	POMAIN			NOGEN/NPR/UNACH						
1	2	109	SYS1	POMAIN			NOGEN/NPR/UNACH						
1	2	115	SYS1	POMAIN	DIU-S2	Q2096-X2	NPR/UNACH/NOAL						
1	2	121	SYS1	POMAIN	STMD	Q2174-X	NPR/UNACH/NOAL						

Step 3: Open Board configuration menu in HiPath 4000 CM Assistant

CM > System Data > Board > Board

Step 4: Add new board

Press **New** button in order to add new board

Step 5: Filling in the mask

- **Part Number:** Q2316-X or Q2316-X10 or Q2324-X500 or Q2324-X510
- **Function ID:** is always 1
- **Category:** IPGW
- **Board Name:** STMI2 or STMI4
- **CGW Function Block:** enter created function block (see [Step 1: Configure function block](#))
- **LTU** and **SLOT:** select free slot (see [Step 2: Search for a free slot](#))

Step 6: Open STMI2-IGW Board Data dialog and fill mandatory fields

Customer LAN IP Address and **Subnet Mask**.

Trunk Protocol: number of b-channels (e.g. **native SIP/SIPQ:** 50).

IMPORTANT: Only one trunk protocol per board is supported! A combination of different trunking protocols on one board is not possible!

After all needed data is entered into required fields press **Save** button.

The board is now added and can be found in **Board Dialog** or in **Board Maintenance**.

7.3.2 Delete IPGW

To delete IPGW board do the following:

- [Step 1: Open Board configuration menu in HiPath 4000 CM Assistant](#)

- [Step 2: Search for IPGW](#)
- [Step 3: Select IPGW and delete it](#)

Step 1: Open Board configuration menu in HiPath 4000 CM Assistant

CM > System Data > Board > Board

Step 2: Search for IPGW

Search Criteria > Category: IPGW

Step 3: Select IPGW and delete it

Choose the board that has to be deleted from the list and click **Delete**. After pressing **Delete** button dialog appears that board is removed from system.

7.3.3 Modify Board Data of IPGW

In order to modify board data follow those steps:

- [Step 1: Open Board configuration menu in HiPath 4000 CM Assistant](#)
- [Step 2: Select IPGW](#)
- [Step 3: Carrying out desired changes](#)
- [Step 4: Checking the changes](#)

Step 1: Open Board configuration menu in HiPath 4000 CM Assistant

CM > System Data > Board > Board

Step 2: Select IPGW

Select **IPGW** in **Category** list and press **Search** button.

Step 3: Carrying out desired changes

Do the needed changes in board dialog (eg. IP address) and save changes with the **Save** button.

After changes are saved dialog about successful action pops up.

Step 4: Checking the changes

Make new board search in order to check if new data is accepted.

7.4 Trunking Between HiPath 4000 and OpenScape Voice with SIP-Q V2

A connection between HiPath 4000 and OpenScape Voice can be established using SIP-Q trunks.

OpenScape Voice provides only a limited range of features over this connection. For details, please refer to the OpenScape Voice documentation.

7.4.1 Restrictions

- Dialing with overlapping digits is not supported in OpenScape Voice (overlapping reception is supported).
- With a HiPath 4000 - OpenScape Voice (transit) - HiPath 4000 scenario, the message waiting display is not routed via the OpenScape Voice.

7.4.2 Configuration HiPath 4000

Please refer to [Section 7.1, "Configuration using AMOs"](#).

The following settings must be configured differently from the configuration given above:

- **Configuration of the gateway number**

In case of a SIP-Q trunking connection to an OpenScape Voice system the parameter **GWDIRNO** in AMO CGWb has to be configured. The value of **GWDIRNO** must be the same as the alias name in the OpenScape Voice system.:

```
CHANGE-  
CGWB:MTYPE=CGW,LTU=1,SLOT=37,TYPE=LEGKDATA,GWNO=1,GWDIRNO=816901;
```

- **Configuration of the external registrar**

In case of a SIP-Q trunking connection to an OpenScape Voice system the HiPath 4000 system must register dynamically with the OpenScape Voice partner system. In this case an external registrar has to be configured.

The external registrar is configured using the AMO CGWB.

- Operation at an external registrar must be activated (**SIPREG=YES**).
- The duration of the registration validity must be entered in **REGTIME**. The partner system registrar data must be entered in the fields **REGIP** and **REPORT**.

```
CHA-  
CGWB:MTYPE=CGW,LTU=<ltu>,SLOT=<slot>,TYPE=SIPTRSSA,SIPREG=  
YES,REGIP1=127.28.45.54,PORTTCP1=5060,REGTIME=<seconds>,RE  
GIP2=0.0.0.0,PORTTCP2=5060;
```

NOTE: OpenScape Voice does not accept values less than 300 seconds in parameter **REGTIME**.

- In AMO CGWB parameter **REGEXTGK** must be set to YES.

```
CHANGE-  
CGWB:MTYPE=CGW,LTU=1,EBT=37,TYPE=LEGKDATA,GWNo=1,GWDIRNO=8  
16901,REGEXTGK=YES;
```

IMPORTANT: Parameters **SIPREG** and **REGEXTGK** must always have the same value.

- Configuration of authentication

If authentication is activated on the OpenScape Voice system the authentication data (SIP user ID in the field **GWUSERID**, password in the field **GWSECRET** and **realm** in the field **GWREALM**) must also be configured for registration with OpenScape Voice. The transmission of authentication data can be activated and deactivated using the parameter **GWAUTREQ**.

```
CHA-  
CGWB:MTYPE=CGW,LTU=<ltu>,SLOT=<slot>,TYPE=SIPTRERH,GWAUTREQ=<  
YES or  
NO>,GWSECRET=<string>,GWUSERID=<string>,GWREALM=<string>;
```

- Block dialing must be set up for the trunking connection to the OpenScape Voice.

7.4.3 Configuration OpenScape Voice

The configuration of the OpenScape Voice is done via the OpenScape Voice Assistant.

In the OpenScape Voice equivalent parameters have to be configured. The following data must be correctly configured:

Signaling and registration parameters that describe the OpenScape Voice registration data (IP address, port, login)

The data entered here must correspond with the data entered in the AMO CGWB for ERH or SSA.

The listening IP address must be entered in the IP address of the external registrar.

The TCP/UDP port must be set in the field **TCP/UDP Port**.

The data set for authentication (**UserID**, **Password**, **Realm**) must correspond with the data entered in the AMO CGWB, in the data branch for SSA.

Aliases:

The gateway ID is used for dynamic registration. This is entered in the AMO CGWB, in the branch **LEGKDATA** in the parameter **GWDIRNO**.

Endpoint data that describes the connection to HiPath 4000

When configuring the endpoint displayed by HiPath 4000 (i.e. HG 3500), the following parameters must correspond with the data configured in the AMO CGWB:

- **Endpoint Name:** This parameter must correspond with the parameter **GWDIRNO** in the AMO CGWB.
- **Registration Type (Static, Dynamic):**
 - If **Static** is entered here, HG 3500 (SIP) registration is not required.
 - If **Dynamic** is entered, HG 3500 (SIP) registration is required.
- **Transport Protocol:** The transport protocol must correspond with the SIP parameter setting in the WBM.
- **Aliases:**

The gateway ID is used for dynamic registration. This is entered in the AMO CGWB in the branch **LEGKDATA** in the parameter **GWDIRNO**.

8 SIP Service Provider

Example configuration

HiPath 4000 system including a native SIP trunk gateway (HG 3500 V4), a set of HiPath 4000 phones and a fax are connected to the HiPath 4000.

The native SIP gateway in HiPath 4000 provides connectivity via the WAN to the SIP service provider. Other SIP phones are registered directly to the service provider. The service provider has a connection to the public PSTN network.

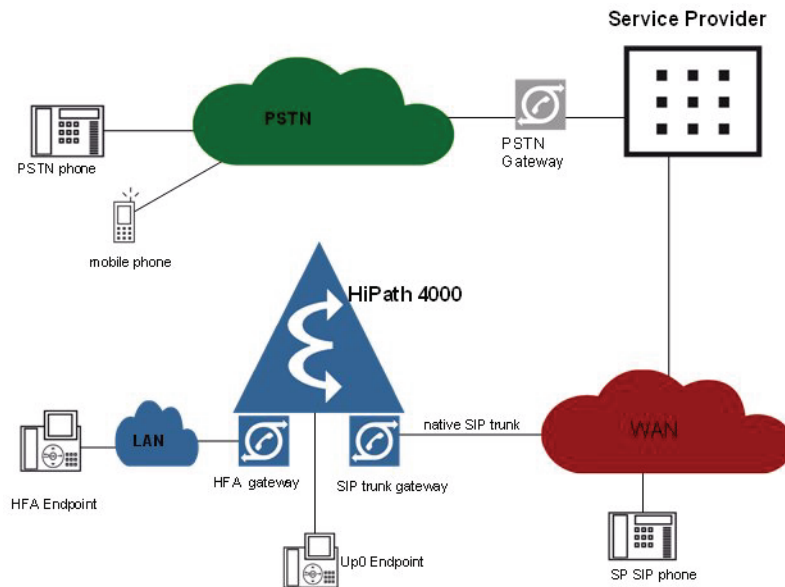


Figure 13

Connectivity to the SIP service provider via WAN and native SIP trunking

8.1 Supported Features

See also [Section 2.1, “Features”](#).

- DNS Client support
- DNS Failover support (Timer T1/T2) / DNS_SRV
- Direct Dial in (DDI) support
- Authentication supported for
 - Gateway device-authentication (REGISTER method) and
 - SIP session-authentication (INVITE method)

8.2 Restrictions

- 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.
- Restrictions of supported features

Feature	Provider	
	T-Systems (version 6.4)	Broadsoft
Display names	yes	yes
Call forwarding	yes	yes
Call transfer (using relInvites)	yes ¹	yes
Call transfer (using Refer)	send: no receive: yes	send: no receive: yes
Message waiting indication supplier	no	yes
TLS	no ²	no

Table 5 Supported features with native SIP trunking partners

- 1 T-Systems only offers one codec, if an INVITE without SDP has been received. This will change with version 7.
- 2 TLS not supported in version 6.4 (planned for version 7)

8.3 SIP Service Provider Configuration

A well configured DNS Server (if possible with DNS_SRV-support) must be available at the SIP-Provider side.

8.4 HiPath 4000 Configuration

8.4.1 General Configuration Rules

IMPORTANT: The HiPath 4000 configuration guidelines are described under the assumption that enough Comscendo licenses for the SIP trunk ports are available and a slot for the STMI2/4-board is reserved in the HiPath 4000 system.

- The configuration covers specific SIP provider data.
- IP addresses
 - The SIP trunk gateway and the service provider must have a fixed IP address that can be reached by each other (no DNS support).
 - Be aware that the IP address range of the HiPath 4000 system (HiPath 4000 Assistant, HFA and SIP phones) are located in the customer LAN and the SIP trunk gateway IP address has an IP address which is routable through the internet. Appropriate protection (firewall) is necessary! For the WBM administration it is necessary to have a https connection to the SIP trunk gateway.
- If not indicated differently all involved parties have CLIP and COLP configured and the session timer of SIP trunk gateway is on (default value).
- The gateway for the SIP service provider connection must be configured as a remote gateway.
- If the SIP service provider is connected through a HiPath 4000 SoftGate a session border controller has to be configured. Please refer to „HiPath 4000 SoftGate“, [Chapter 6, “Installing the “Comdasys Convergence 1600” Session Border Controller”](#).

8.4.2 Trunk Configuration

AMO TDCSU:

```
ADD-TDCSU:OPT=NEW,PEN=1-01-037-  
0,COTNO=36,COPNO=32,DPLN=0,ITR=0,COS=2,LCOSV=1,LCOSD=1,CCT="GW2  
",PROTVAR="ECMAV2",SEGMENT=8,ISDNIP=00,ISDNPN=0,TRACOUNT=31,SATC  
OUNT=MANY,NNO=1-69-  
149,ALARMNO=12,COTX=36,CCHDL=SIDEA,CLASSMRK=EC&G711,TCCID="IP  
RS",TGRP=50,SRCHMODE=DSC,INS=Y,DEV=HG3550CO,BCHAN=1&&30,BCNEG=N,  
BCGR=1,LWPAR=0;
```

Besides AMO BFDAT, AMO BCSU, AMO BUEND and AMO TDCSU (ECMAV2 protocol) AMO COT and AMO COP are of interest.

Example:

```
ADD-  
COT:COTNO=80,PAR=ANS&CEBC&BSHT&BLOC&LWNC&NLCR&TSCS&DFNN&NLRD&NOF  
T&NTON;  
ADD-COP:COPNO=80,TRK=TA,TOLL=TA;
```

Routing and numbering plan specific issues

Routing and numbering plan specific issues see below (AMO TDCSU, AMO COT, AMO COP).

CCT

Enter circuit from gateway 2.

8.4.3 Gateway Board Configuration

IMPORTANT: Gateway loadware version pzksti40.os.o4.019-003 or higher should be used.

HG 3500 supports DNS/SRV and uses SIP trunking profiles. Therefore all provider specific data has to be configured via WBM (see [Section 7.2](#), "Configuration using SIP Trunk Profiles in the WBM").

AMO CGWB (in the following example with **IPADR=212.202.240.122**) is used to configure the used trunk protocol (native SIP), DNS/SRV server (used by profile) and the codec list (for example VAD), etc.

The gateway is configured as local gateway with AMO GKREG only for INTGW without any provider specific data (there is no difference at AMO CGWB / AMO LDAT configuration regarding NON-registration and registration).

Example of Service Provider Configuration

```
ADD-
CGWB:LTU=1, SLOT=8, SMODE=NORMAL, IPADR=212.202.240.122, NETMASK=255.255.255.248;

CHANGE-
CGWB:MTYPE=CGW, LTU=1, SLOT=8, TYPE=GLOBIF, PATTERN=213, VLAN=NO, VLAN ID=0, DEFRT=212.202.240.121, BITRATE="AUTONEG", TRPRSIP=60, TRPRSIPQ=0, TRPRH323=0, TPRH323A=0, TLSP=4061, DNSIPADR=212.202.240.121;

CHANGE-CGWB:MTYPE=CGW, LTU=1, SLOT=8, TYPE=SERVIF, LOGINTRM="TRM";

CHANGE-
CGWB:MTYPE=CGW, LTU=1, SLOT=8, TYPE=ASC, UDPPRTLO=29100, UDPPRTHI=29339, TOSPL="184", TOSSIGNL="104", T38FAX=YES, RFCFMOIP=YES, RFCDTMF=YES, REDRFACTN=NO, PRIO=PRIO1, C

ODEC=G729AB, VAD=YES, RTP="20";

CHANGE-
CGWB:MTYPE=CGW, LTU=1, SLOT=8, TYPE=ASC, PRIO=PRIO2, CODEC=G729A, VAD=NO, RTP="20";

CHANGE-
CGWB:MTYPE=CGW, LTU=1, SLOT=8, TYPE=ASC, PRIO=PRIO3, CODEC=G711A, VAD=YES, RTP="20";

CHANGE-
CGWB:MTYPE=CGW, LTU=1, SLOT=8, TYPE=ASC, PRIO=PRIO4, CODEC=G711U, VAD=YES, RTP="20";
```

```

0";
CHANGE-
CGWB:MTYPE=CGW,LTU=1,SLOT=8,TYPE=ASC,PRIO=PRIO5,CODEC=G723,VAD=YES,RTP="30
";
CHANGE-
CGWB:MTYPE=CGW,LTU=1,SLOT=8,TYPE=ASC,PRIO=PRIO6,CODEC=NONE,VAD=YES,RTP="20"
;
CHANGE-
CGWB:MTYPE=CGW,LTU=1,SLOT=8,TYPE=ASC,PRIO=PRIO7,CODEC=NONE,VAD=YES,RTP="20"
;
CHANGE-
CGWB:MTYPE=CGW,LTU=1,SLOT=8,TYPE=DSP,VADEN=NO,JITBUFD="60";
CHANGE-
CGWB:MTYPE=CGW,LTU=1,SLOT=8,TYPE=LEGKDATA,GWNO=11,GWDIRNO=0,REGEXTGK=NO;
CHANGE-CGWB:MTYPE=CGW,LTU=1,SLOT=8,TYPE=SIPTRERH,GWAUTREQ=NO;
CHANGE-
CGWB:MTYPE=CGW,LTU=1,SLOT=8,TYPE=SIPTRSSA,SIPREG=NO,REGIP1=0.0.0.0,PORTTCP1=5060,PORTTLS1=5061,REGTIME=120,REGIP2=0.0.0.0,PORTTCP2=5060,PORTTLS2=5061;
CHANGE-
CGWB:MTYPE=CGW,LTU=1,SLOT=8,TYPE=DLSDATA,DLSSIPADR=192.168.10.68,DLSPORT=10444,DLSPAS=NO;
CHANGE-
CGWB:MTYPE=CGW,LTU=1,SLOT=8,TYPE=JB,AVGDLYV=40,MAXDLYV=120,MINDLYV=20,PACKLOSS=4,AVGDLYD=60,MAXDLYD=200,JBMODE=0;
ADD-
GKREG:GWNO=11,GWATTR=INTGW&HG3550V2&SIP,DIPLNUM=0,DPLN=0,LAUTH=1,INFO="LOCAL GW";
ADD-LDAT:LROUTE=300,LSVC=ALL,LVAL=1,TGRP=1,ODR=300,LAUTH=1;

```

8.4.4 Routing and Numbering Plan Configuration

For outgoing calls block dialing has to be configured for SIP provider trunk, dialing pattern has to end with e.g. '-Z'.

```

ADD-LDPLN:LCRCONF=LCRPATT,DIPLNUM=0,LDP="0"-
"Z",LROUTE=1,LAUTH=1;

```

LCR configuration rules depend on the numbering plan, that is used between HiPath 4000 and SIP service provider.

The preferred numbering plan from our side is **NPI= ISDN, TON= INTERNATIONAL** (explicit format). Implicit numbering plan may cause more configuration depending on in which format the provider expects the called and calling party.

Hints concerning AMOs

AMO COT: Parameter **LINC** is added to define calling number as an implicit ISDN number without access codes.

AMO COP: Parameter **IMEX** is added to define called number as an implicit ISDN number.

AMO KNFOR: specifies calling party modification for implicit national or international numbering plan.

AMO TDCSU: **ISDNIP=00**, **ISDNNP=0** specifies national and international prefix for implicit ISDN number for CO trunking.

AMO SDAT: Be aware of the used numbering plan when using **PUBNUM**.

9 SIP Subscriber

The feature “SIP Subscriber for HiPath 4000” encompasses the operation of SIP endpoints, which can be run on a HiPath 4000 system using a HG 3500 board.

The operation of SIP endpoints on IPDA shelves is also supported.

The following SIP endpoints are supported:

- OpenStage SIP family
- optiPoint 410 (without Entry),
- optiPoint 420 family,
- optiPoint 150 S,
- optiClient 130 S and
- AP1120.

In terms of subscriber support, the operation of HFA stations that run on an HG 3500 board is possible.

In both call directions, this feature encompasses

- G.7xx audio calls
 - from SIP to SIP
 - from SIP to HFA
 - from SIP to TDM
- G.711 transparent FAX/MODEM calls
 - from SIP to SIP
 - from SIP to HFA
 - from SIP to TDM

9.1 Service Information

- Up to 120 SIP endpoints can be configured on a HG 3500 board, these can be spread across all available ports or configured on one port.
- RFC 2833 (DTMF via RTP) and RFC 2198 (Support of Redundant Audio Data) are supported.
- The following table shows the combinations of the parameters **SECZONE**, **USERID** and **PASSWD** in AMO SBCSU to make secure registration possible.

Parameter			
SECZONE	USERID	PASSWD	Possible
Set	Set	Set	Yes
Not Set	Set	Set	Yes
Set	Not Set	Set	Yes
Not Set	Not Set	Set	Yes
Set	Not Set	Not Set	No
Not Set	Set	Not Set	No
Set	Set	Not Set	No
Not Set	Not Set	Not Set	Yes

- **For optiPoint 410 and 420**

The configuration and loading of software, amongst others, is possible via the WBM or the DLS.

Please refer to the administration manual of optiPoint 410/420 SIP for WBM issues (http://apps.g-dms.com:8081/edoku/jsp/searchresult_v2.jsp?edokutype=&search_mode=term&product=&product_version_main=&product_version_sub=&search_term_type=all&term=optiPoint%20410/420%20S%20HiPath%204000&sort_result=title&docclass=user&language=en&checkdate=&lang=en).

DLS V2.0: <http://apps.g-dms.com:8081/techdoc/en/P31003S2320M1000176A9/index.htm>

- **For optiPoint 150 S**

For information on configuring and SIP settings please refer to the administration manual of optiPoint 150 S (http://apps.g-dms.com:8081/edoku/jsp/searchresult_v2.jsp?edokutype=&search_mode=product&product=optiPoint%20150%20S&product_version_main=&product_version_sub=&search_term_type=all&term=&sort_result=title&docclass=service&language=en&checkdate=&lang=en)

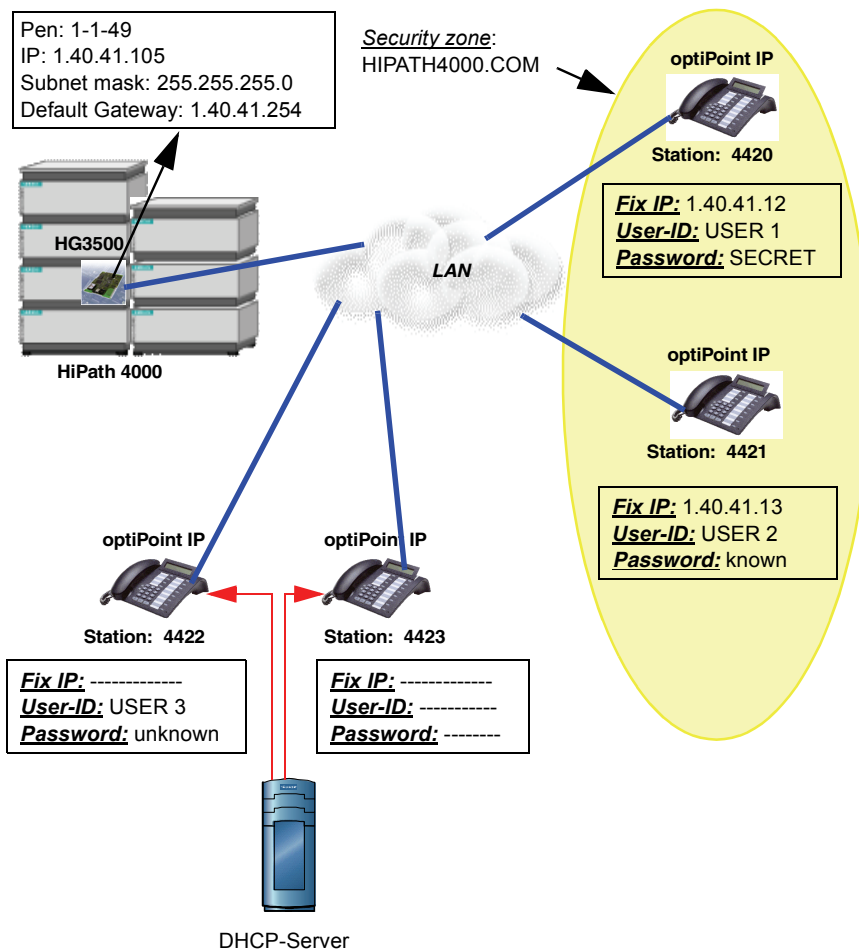
9.2 Restrictions

- For codec G.729 there is one restriction.

The WBM interface allows you to configure all codecs from the G.729 family (G.729, G.729A, G.729B and G.729.AB). Only the highest priority codec is used for SIP signaling.


- SIP subscribers cannot be a member of an ONS group.

9.3 Configuration



9.3.1 STMI2/4 Board Configuration

	<p>Configuration Management --> System Data --> Board --> CGW Function Block Click New, enter data and click Save.</p> <p>Configuration Management --> System Data --> Board --> Board Click New, enter data (header information, General Board Data tab, STMI2-IGW Board Data tab, CGW Functionalities tab) and Save.</p>
--	---


	<pre>ADD-BFDAT:FCTBLK=1,FUNCTION=SIP,BRDBCHL=BCHL60; CHANGE- BFDAT:CONFIG=CONT,FCTBLK=1,FUNCTION=SIP,LINECNT=10,BCHLCN T=10; CHNAGE-BFDAT:CONFIG=OK,FCTBLK=1,ANSW=YES; ADD-BCSU:MTYPE=IPGW,LTG=1,LTU=1,SLOT=49,PARTNO="Q2316-X ",FCTID=1,FCTBLK=1; ADD- CGWB:LTU=1,SLOT=49,SMODE=NORMAL,IPADR=1.40.41.105,NETMASK =255.255.255.0,DEFRT=1.40.41.254; The Parameter GWAUTREQ and SIPREG must be set to NO,in situations where the board is used for SIP Subscriber. CHANGE-CGWB:MTYPE=CGW,TYPE=SIPTRERH,GWAUTREQ=NO; CHANGE-CGWB:MTYPE=CGW,TYPE=SIPTRSSA,SIPREG=NO;</pre>
---	---


9.3.2 Configuring a SIP Subscriber

IMPORTANT: Configuration of HG 3500 (CGW) board is prerequisite for SIP subscriber support (see [Section 9.3.1, "STMI2/4 Board Configuration"](#)).



IMPORTANT: If the parameters **SECZONE**, **USERID** and **PASSWD** are used for a secure registration in the AMO SBCSU, the parameter **AUTHERF** is set to **YES** and the security data must be set also in the SIP terminal !

IMPORTANT: The parameter **MBCHL** in AMO SDAT should be set for all SIP subscribers. If not features, which need a second b channel are not possible (e.g. conference, second call).

	<p>Configuration Management --> Station --> Station Click New. Enter Device Family and Device Combination: S0PP. Enter Connection Type: SIPSEC. Choose a PEN that has the Board Type: STMI2IGW. Enter all COS* and LCRCOS* values on Basic 1 tab sheet. Choose S0PP/S2PP Protocol on Bus Extension tab sheet. Click Save.</p>
---	---



	<p>ADD-SBCSU:STNO=4420, OPT=FPP, CONN=SIP, PEN=1-1-49-0, DVCFIG=S0PP, COS1=1, COS2=1, LCOSV1=1, LCOSV2=1, LCOSD1=1, LCOSD2=1, PROT="SBDSS1", OPTIDX=10, IPADDR=1.40.41.12, PASSWD="SECRET", USERID="USER 1", SECZONE="HIPATH4000.COM", FIXEDIP=YES, AUTHREQ=YES, SMGSUB=NO;</p> <p>CHANGE-SDAT:STNO=4420, TYPE=ATTRIBUT, AATTR=MBCHL;</p>
---	---

9.3.3 Removing a SIP Subscriber

	<p>Configuration Management --> Station --> Station Click Search. Select subscriber you want to delete. Click Delete.</p>
	<p>DELETE-SBCSU:STNO=4420, SVC=ALL;</p>



9.3.4 Changing a SIP Subscriber

The SIP endpoint data is changed under branch "OPT=FPP"

	<p>Configuration Management --> Station --> Station Click Search. Select subscriber you want to change. Change data. Click Save.</p>
	<p>CHA-SBCSU:STNO=4420, OPT=FPP, IPADDR=1.40.41.15, FIXEDIP=YES;</p>

9.3.5 Enabling/Disabling DMC

DMC is enabled for station 4420 and disabled for station 4422 and 4423.

	<p>Configuration Management --> Station --> Station Click Search. Select subscriber you want to change the DMC settings. Activate/deactivate check box DMC allowed on Basic 3 tab sheet.</p>
	<p>CHANGE-SDAT:STNO=4420, TYPE=ATTRIBUT, AATTR=DMCALLWD;</p>

9.3.6 Call Forwarding

3 kinds of call forwarding are possible:

- CFU - Call Forwarding Unconditional

```
ADD-ACTDA:TYPE=STN,STNO=<station
number>,FEATCD=FWD,CFVAR=<call forwarding
variant>,DTYPE=CFU,SI=VCE;
```

- CFNR - Call Forwarding No Reply

```
ADD-ACTDA:TYPE=STN,STNO=<station
number>,FEATCD=FWD,CFVAR=<call forwarding
variant>,DTYPE=CFNR,SI=VCE;
```

- CFB - Call Forwarding Busy

```
ADD-ACTDA:TYPE=STN,STNO=<station
number>,FEATCD=FWD,CFVAR=<call forwarding
variant>,DTYPE=CFB,SI=VCE;
```

9.3.7 Second Call

The attribute **MBCHL** (multiple b channels) must be set.

```
CHANGE-SDAT:STNO=<number of the sip
subscribers>,TYPE=ATTRIBUT,AATTR=MBCHL;
```

9.3.8 Relevant AMOs

AMO	Parameter	Sprache/ Language	Beschreibung/ Description
ACTDA	UART	d	CFB: Anrufumleitung bei besetzt CFNR: Anrufumleitung bei nicht melden CFU: ständige Anrufumleitung
	DTYPE	e	CFB: call forwarding busy CFNR: call forwarding no reply CFU: Call forwarding unconditional
BCSU	ALARMNR	d	Alarm Nummer
	ALARMNO	e	Alarm number
	ART=HWYBDL	d	Highway-Bündel
	TYPE=HWYBDL	e	Highway bundle
	BKANSIP	d	Anzahl der B-Kanaele fuer die SIP- Subscriber Funktion
	BCHLSIP	e	Number of b channels for sip subscriber function
	FCTID	d	Function Id (wird immer auf 1 gesetzt)

AMO	Parameter	Sprache/ Language	Beschreibung/ Description
	FCTID	e	Function id (is always set to 1)
	FCTBLK	d	Funktionsblock-Index (einen beliebigen freien Funktionsblock zwischen 1-20 wählen)
	FCTBLK	e	Function block index
	LWVAR	d	Index auf Loadware Block der T1 Baugruppe
		e	Loadware variant
	SACHNR	d	Baugruppensachnummer (2. und 3. Block) Q2316-X, Q2316-X10, Q2324-X500, Q2324-X510
	PARTNO	e	Part numver (2nd and 3rd bloc) Q2316-X, Q2316-X10, Q2324-X500, Q2324-X510
	TYP=IPGW	d	IP Gateway (Common Gateway Baugruppe)
	MTYPE=IPGW	e	IP Gateway (Common gateway board)
BFDAT	ANZSATZ	d	Anzahl der funktionsbezogenen Saetze. Mögliche Werte: 1-240
	LINECNT	e	Defines the number of lines related to the selected function.
	BGBKAN	d	Block fuer Baugruppe mit 60 und/oder 120 B-Kanaelen
	BRDBCHL	e	dedicates the block for boards with 60 and/or 120 b-channels
	CONFIG=WEITER	d	Weitere Block-Konfiguration ermöglichen
	CONFIG=CONT	e	Continue block configuration
	CONFIG=OK	d	Block-Konfiguration abschließen
	CONFIG=OK	e	Finish block configuration
	FCTBLK	d	Dieser Index beschreibt den Funktionsblock welcher auf dem Common Gateway konfiguriert werden soll. Anhand des Funktionsblocks wird die Konfiguration der benötigten pyhsikalischen Lines (Sätze der Baugruppe) festgelegt.
	FCTBLK	e	This index describes the function block which should be configured on the common gateway board. With that index the amount of needed physical lines (board circuits) is calculated.

AMO	Parameter	Sprache/ Language	Beschreibung/ Description
	FUNCTION	d	Dieser Parameter legt das Konfigurationsprofile des Common Gateways fest. Dabei muss die eventuell benötigte HG 3570 Funktion als erste angeführt werden. Falls ein bestimmter Line-Bereich für die Funktionen HG 3530 oder HG 3550 vorreserviert werden soll, muss die entsprechende Funktion am Ende stehen und mit dem Wert HG35xxR abgeschlossen sein. Die Funktion STANDBY kann nur als Einzel-Funktion konfiguriert werden. FUNCTION=SIP: SIP Subscriber Funktion
	FUNCTION	e	This parameter defines the configuration profile of the common gateway board. If HG3570 functionality is used, it must be configured at first position. If a prereservation of a certain line range of functions HG3530 or HG3550 is desired, this function must be at the end of the profile just suffixed by the according HG35xxR value. The function STANDBY can only be configured as single function. FUNCTION=SIP:SIP subscriber function
CGWB	DEFRT		IP-Adresse des Default Routers innerhalb des LAN-Segmentes. Der Default Router übernimmt die Weiterleitung der Pakete, die eine Zieladresse außerhalb des eigenen LAN-Segmentes haben. 0.0.0.0 bedeutet, daß kein Default Router konfiguriert ist.
	DEFRT	e	IP address of the default router within the LAN segment. The default router takes care of routing forward all packets with a destination address with a network part different from the own LAN segment. 0.0.0.0 means that no default router is configured.
	IPADR	d	IP Adresse der Common Gateway Baugruppe (Source Adresse)
	IPADR	e	IP address of common gateway board (source address)
	MTYP=INITCGWB	d	Zurücksetzen der Konfigurationsdaten der Common Gateway Baugruppe auf die Standardwerte
	MTYPE=INITCGWB	e	Reset configuration data of common gateway board to default values

AMO	Parameter	Sprache/ Language	Beschreibung/ Description
	NETMASK	d	IP-Netzmaske des LAN-Segmentes. Die IP-Netzmaske bestimmt die Grenze zwischen Netz- und Host-Teil in der IP-Adresse. Alle IP-Adressen am LAN-Segment müssen bezüglich des Netzanteils der IP-Adresse gleich und bezüglich des Host-Teils unterschiedlich sein (auch der Default Router muss dieser Bedingung entsprechen).
	NETMASK	e	IP net mask of LAN segment The IP net mask determines the network and the host partition of an IP address. All IP addresses of a LAN segment must contain the identical network addresss part but different host address parts (also the Default Router must fulfill this requirement).
	SMODE=NORMAL	d	Standby Mode oder Normal Mode Eine Baugruppe im Normal Mode hat gültige Baugruppendaten und normalerweise auch OPTIIPs konfiguriert. Eine Baugruppe im Standby Ready Mode hat keine gültigen Baugruppendaten, auf diese Baugruppe können OPTIIPs umgeschaltet werden, falls eine andere Baugruppe aus demselben Baugruppen-Pool (AMO BPOOL) defekt wurde. Eine Baugruppe im Standby Defekt Mode hat ebenfalls keine gültigen Baugruppendaten, diese Baugruppe hat aufgrund eines Defekts seine OPTIIPs und seine Baugruppendaten abgegeben.
	SMODE=NORMAL	e	Standby Mode or Normal Mode A board in Normal Mode has valid board data and normally also OPTIIPs assigned to it. A board in Standby Ready Mode has no valid board data, to this board OPTIIPs can be switched over if another board of the same board reconfiguration pool (AMO BPOOL) becomes defective. A board in Standby Defect Mode has also no valid board data, this board has lost its OPTIIPs and its board data to another board because it's gone defective.
SBCSU	ART=FPP	d	Haupttrufnummer des S0-/S2-Punkt zu Punkt Anschlusses
	OPT=FPP	e	Functional point-to-point connection
	ANSCHL=SIP	d	Anschlussart=SIP
	CONN=SIP	e	Kind of connection=SIP
	GERKON=S0PP	d	S0-Punkt zu Punkt Anschluss

AMO	Parameter	Sprache/ Language	Beschreibung/ Description
	DVCFIG=S0PP	e	S0-point-to-point
	PROT=SBDSS1	d	Protokollvariante DSS1
	PROT=SBDSS1	e	Protocol variant DSS1
	OPT=10	d	Optiontabellenindex 10
	OPTIDX=10	e	Option table index 10
	IPADR	d	IP-Adresse
	IPADDR	e	IP address
	PASSWD	d	Passwort
	PASSWD	e	Password
	KENNUNG	d	Benutzerkennung
	USERID	e	User ID
	SECBER	d	Security-Bereich (Art Domäne)
	SECZONE	e	Security zone (kind of a Domain)
	FESTEIP	d	Feste IP-Adresse
	FIXEDIP	e	Fixed IP address
	AUTHERF	d	Authentifizierung erforderlich
	AUTHREQ	e	Authentication required
	SMGTLN	d	SMG Teilnehmer; Für Survivability-Lösung mit der H8k;
	SMGSUB	e	SMG subscriber For survivability solution with H8k;
SDAT	EMERK	d	Teilnehmermerkmal einfügen
	AATTR	e	Add attribute
	AMERK	d	Teilnehmermerkmal ausfügen
	DATTR	e	Delete attribute

9.4 optiPoint 410/420

The terminal settings can be done via the configuration menu at the terminal or via its WBM (Web Based Management) or via the DLS (Deployment Service).

In this document only the basic steps via the WBM and via the DLS of the optiPoint SIP software of the terminal are described. For detailed information please use the Administration respectively User manual.

Administration and user manual:

http://apps.g-dms.com:8081/edoku/jsp/searchresult_v2.jsp?edokutype=&search_mode=term&product=&product_version_main=&product_version_sub=&search_term_type=all&term=optiPoint%204

[10/](#)

[420%20S%20HiPath%204000&sort_result=title&docclass=user&language=en&checkdate=&lang=en](#)

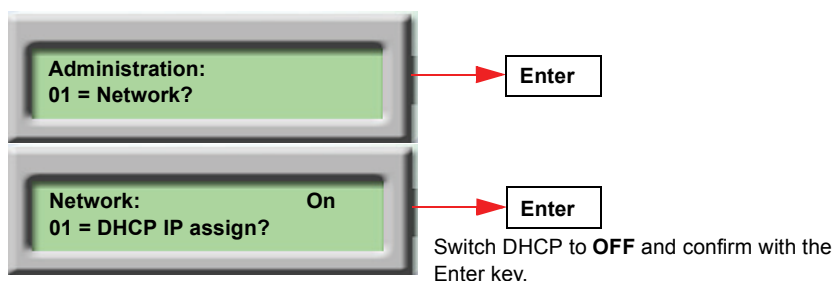
DLS:

<http://apps.g-dms.com:8081/techdoc/en/P31003A2056L1240176A9/index.htm>

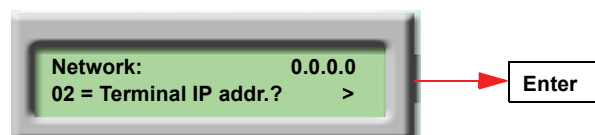
9.4.1 Required settings for the terminal without DHCP

If the customer has no DHCP server, DHCP must be switched off at the terminal and the appropriate IP data have to be entered.

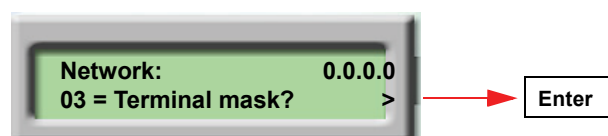
Change to **05=Setup** -> Press key 6 at the number block -> Enter the Administration Password "123456" (standard)



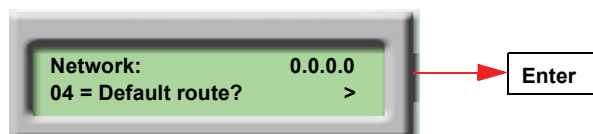
Change to **02 = Terminal IP addr.:**



Change to **03 = Terminal mask:**



Change to **04 = Default route:**



Change to **06 = QoS:**



The further settings are described in this documentation via the WBM of the terminal and via the DLS. They are also valid using DHCP.

9.4.2 Settings via the WBM of the terminal

Procedure: Administration -> System -> SIP Environment

- Terminal details:
 - **Phone number:** Subscriber number (parameter **STNO**) from **AMO SBCSU**.
- SIP details:
 - **SIP routing** must be set to **Server**.
 - **Registrar IP address or DNS name** and **Server IP address or DNS name:**
Source IP address of the HG 3500 from **AMO CGWB**, parameter **IPADR**.
 - **SIP port** must be set to **5060**.
 - **RTP Base port** must be set to **5004**.
 - **SIP transport** must be set to **UDP**.
 - **SIP server type** must be to **Other**.
 - **SIP realm, SIP user ID, New SIP password, Confirm SIP password:**
These parameters are set, if security data for the registration on the HG 3500 is used.
SIP realm: Entry in parameter **SECZONE** in **AMO SBCSU**.
SIP user ID: Entry in parameter **USERID** in **AMO SBCSU**.
New SIP password, Confirm SIP password: Entry in parameter **PASSWD** in **AMO SBCSU**.

Procedure: Administration -> Speech

- **Codec:** G.729
- **Audio mode:** Low Bandwidth Preferred
- **RTP packet size:** 20ms
- Checkboxes **Silence Suppression** and **Microphone Disable** are not checked.

Procedure: Administration -> Configuration Management... -> Settings

If a Deployment Service Server (DLS) is used and this server IP address is not set via the DHCP server, the IP address must be set via the WBM for the terminal.

Procedure: Administration -> Quality of Service

Depending on the customer situation, the Quality of Service settings must be performed (is possibly already be set in the terminal via the configuration menu without DHCP).

Standard settings are: L2 and L3 are set to ON.

	Layer 2	Layer 3
Required	checked	checked
Voice	5	EF
Signaling	3	Af31
Default	0	---
VLAN Discovery	Dhcp	
Manual VLAN id	0	

9.4.3 Settings via the Deployment Service (DLS)

9.4.3.1 Scanning workpoints

Main Menu > Workpoints > Workpoint Interaction > Scan Workpoints

This function lets a DLS user create a DLS database of all workpoints in the network for processing purposes. Workpoints do not need to be registered at the DLS in this case.

IP Ranges tab

Use this tab to specify an IP address range/port number combination for the workpoints to be scanned. You can add another combination with the button **New**. In addition, you can import a CSV file containing IP/port combinations and export this data in CSV format.

Configuration tab

- **Send DLS Address** checkbox:

Use the DLS on an TAP to activate the **Send DLS Address** check box and identify the DLS address and the DLS port number on the EWS. During scanning, the workpoints are informed of the address data of the DLS that serves them.

IMPORTANT: The Send DLS Address option must not be used if there is a permanent DLS server in the network.

- **DLS Address**

IP address of the DLS server on the EWS.

Format: 000.000.000.000, 000 = value between 000 and 255

- Select **Save**.
- Select **Scan Workpoints**:
Starts the IP scanner displayed in the Object view.

9.4.3.2 Setting Workpoint Configuration

Same settings as in [Section 9.4.2, "Settings via the WBM of the terminal"](#).

Main Menu > Workpoints > optiPoint Configuration > Gateway / Server / DLS

Check information on **Gateway** and **SIP Registering 1** tab.

SIP Registering 2 tab

- **SIP Realm, SIP User ID, SIP Password:**

These parameters are set, if security data for the registration on the HG 3500 is used.

SIP Realm: Entry in parameter **SECZONE** in **AMO SBCSU**.

SIP User ID: Entry in parameter **USERID** in **AMO SBCSU**.

SIP Password: Entry in parameter **PASSWD** in **AMO SBCSU**.

Main Menu > Workpoints > optiPoint Configuration > Audio Settings

Audio Settings tab

- **Codec:** Low bandwidth preferred
- **Compression:** G.729

- **Packet Size: 20mS**

Main Menu > Workpoints > optiPoint Configuration > Quality of Service

Quality of Service settings on **QoS Parameter** tab.

9.5 optiPoint 150 S

The settings for the terminal can be done via the configuration menu of the phone Menu or via the WBM (Web Based Management).

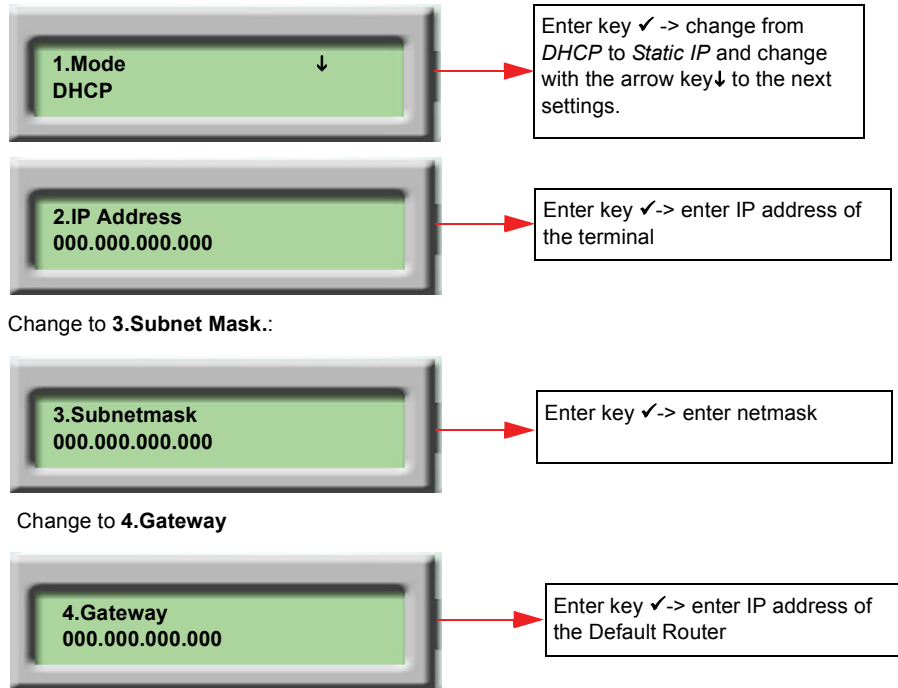
IMPORTANT: The Deployment Service (DLS) does not support the optiPoint 150 S terminal!

In this document only the basic steps via the WBM of the new optiPoint SIP software of the terminal are described. For detailed information please use the Administration respectively User manual (http://apps.g-dms.com:8081/edoku/jsp/searchresult_v2.jsp?edokutype=&search_mode=product&product=optiPoint%20150%20S&product_version_main=&product_version_sub=&search_term_type=all&term=&sort_result=title&docclass=&language=en&checkdate=&lang=de).

9.5.1 Required settings for the terminal without DHCP

If the customer has no DHCP server, DHCP must be switched off at the terminal and the appropriate IP data have to be entered.

Menu key ■ -> change with the arrow key ↓ to Menu **5.Settings** -> Enter key ✓ -> with arrow key ↓ to Menu **2.Network** -> "123456" -> **1.Mode** -> with Enter key ✓



IMPORTANT: QoS settings can **only** be done via the WBM of the terminal !

The further settings are described via the WBM in this documentation. They can be also used with DHCP (standard setting).

9.5.2 Settings via the WBM of the Terminal

Access to the WBM of the terminal

To invoke the interface, open a Web browser and enter the following URL:

http://[telephone IP]

[telephone IP] is the IP address of the optiPoint 150 S you want to map, e.g.:
http://1.40.11.14.

SIP settings

SIP Setup > SIP Settings

- **Registrar IP address or DNS name (SIP), Server IP address or DNS Name (SIP):**
Source IP address of the HG 3500 from **AMO CGWB**, parameter **IPADR**.
- **Port: 5060**
- **Phone Number:** Station number (parameter **STNO**) from **AMO SBSCSU**.
- **SIP user ID, New SIP password, Confirm SIP password:**
These parameters are set, if security data for the registration on the HG 3500 is used.
SIP user ID: Entry in parameter **USERID** in **AMO SBSCSU**.
New SIP password, Confirm SIP password: Entry in parameter **PASSWD** in **AMO SBSCSU**.

IMPORTANT: SIP realm: Entry in parameter **SECZONE** in **AMO SBSCSU**.
This parameter cannot be used for optiPoint 150 S!

Check and/or set the SIP extension settings

SIP Setup > SIP Extensions

Codec settings

Phone Setup > Codec

Language, date and time settings

User Setup > Localization

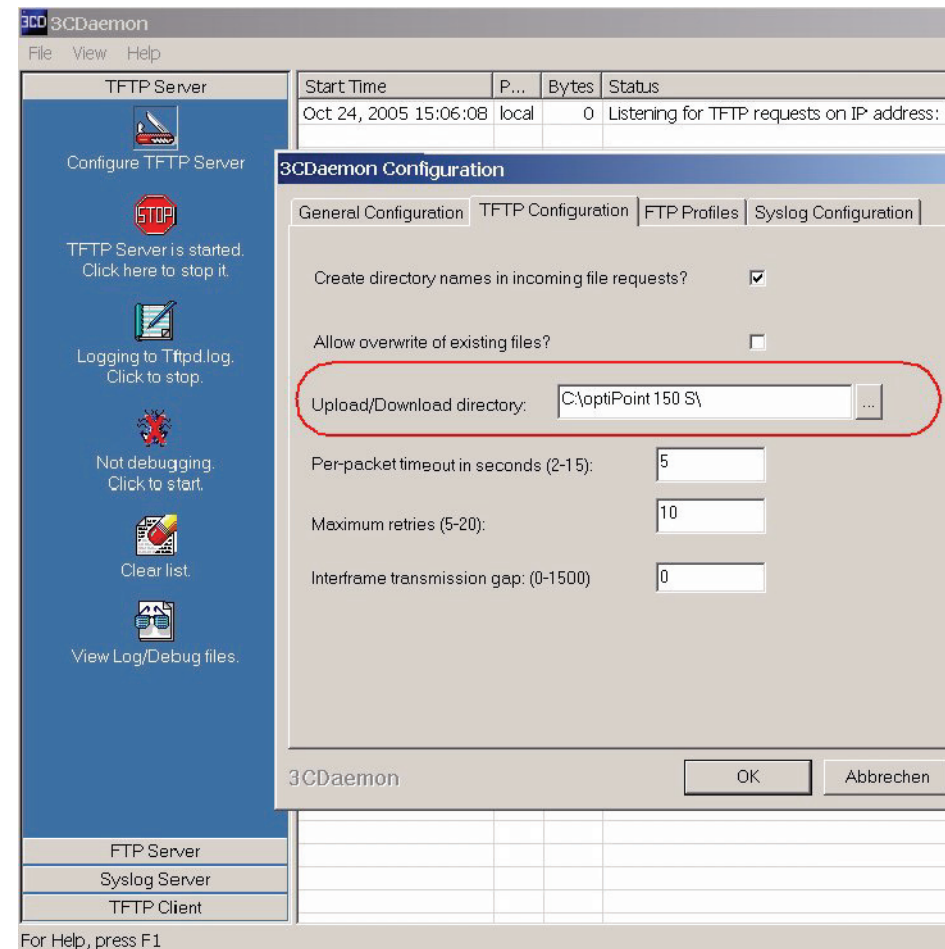
Backup the configuration- and the phone book data

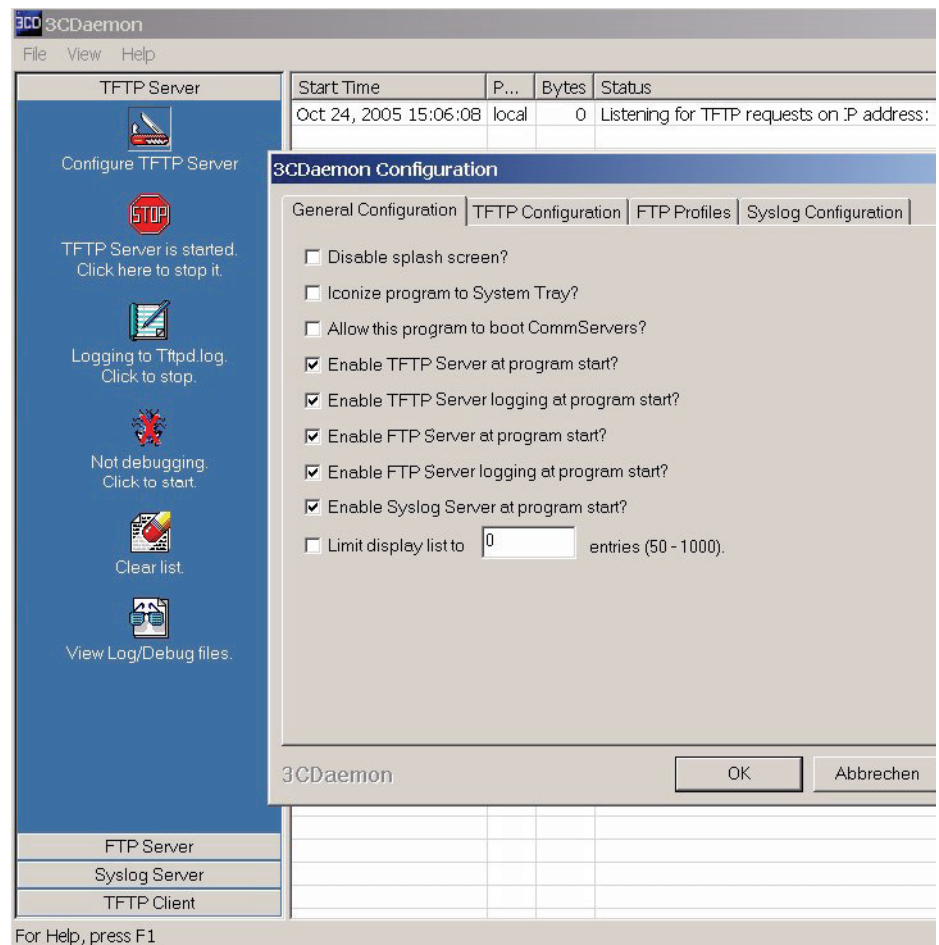
Utilities > Backup Settings

9.5.3 Software Supply for optiPoint 150 S

In this example the TFTP server 3C Daemon is used.

Configure TFTP server





Settings for the software upgrade in the WBM of optiPoint 150 S

Utilities > Upgrade

9.6 optiClient 130 S

optiClient 130 is a PC-based multimedia application offering connection services from different communication media via LAN (network). Voice, video, and chat connections (instant messages) can be administered and managed with optiClient 130. optiClient 130 is operated like a phone on your PC.

For more information please refer to administrator and user documentation:

http://apps.g-dms.com:8081/edoku/jsp/searchresult_v2.jsp?edokutype=&search_mode=product&product=optiClient%20130%20S&product_version_main=&product_version_sub=&search_term_type=all&term=&sort_result=title&docclass=&language=en&checkdate=&lang=en

9.6.1 Installation and Settings

In this document only the basic steps are described. For detailed information please use the Administration /User manual.

The installation will be started with the **Setup.exe** file. Then follow the menu of the installation routine.

During the installation routine, it is possible to select a **Standard Provider** and the **DLS Server** data.

After the optiClient 130 S software is installed and started on the PC, the following settings and entries have to be carried out.

Starting optiClient

The installation routine creates a program group and shortcut for optiClient 130 on the desktop.

To start optiClient 130

- double-click the optiClient 130 shortcut icon on the desktop
- or
- select **Start > Program Files > Siemens > optiClient**.

Logon

The Logging On mask appears. If there is no password saved for the last user, logon is automatically performed with the last settings entered (user, language, location) when optiClient 130 is started). If you want to force the logon mask to appear, for example, to enter another user, hold the Shift-key pressed when starting the optiClient 130 program.

Log on to optiClient 130 as a user first when the program starts:

Enter your **Login**, **Password** and **Location** and select a **Language**.

Different administration functions are available for managing the user or site information and for switching to the configuration - depending on the settings currently configured. Click **Manage** for this.

Settings menu

Manage > Settings

Opens the Settings dialog for configuring the optiClient 130 parameters. Once you have finished editing your settings you are returned to the Logon dialog. The current settings are applied when you log on.

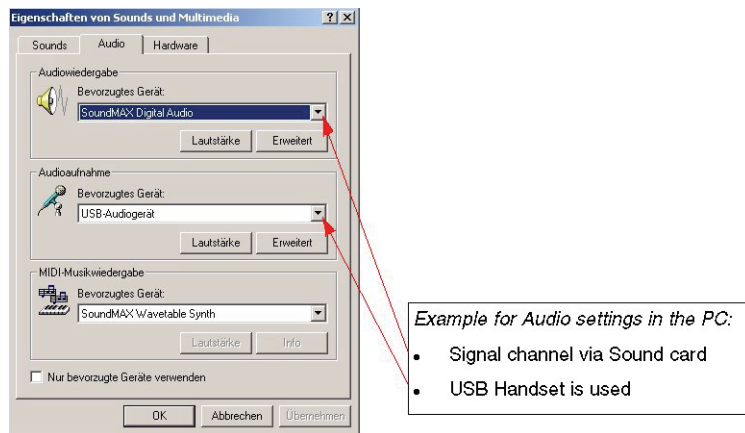
- Color scheme

General > General

- Audio schemes

User Interface Modules > Settings for sound control

The Sound control module is responsible for audio control (audio buttons, volume menu, volume control, additional speaker button) in your optiClient 130. No additional settings are needed for the module.



- Options for click

User Interface Modules > Phone > Integrated phone

In this menu it is possible to define the options for click in the display of the Client:

– Display integrated phone as popup window

If you click the display, the dialing and dialog keys appear on the display as a pop-up window (as individually set). The keys can then be operated as usual in the window. The pop-up window containing the keys disappears as soon as you click another window.

– Display free phone

If you click the integrated phone's display, the free phone opens (as individually set). If the free phone is already open, you switch to this window.

– Show and hide free phone

If you click the integrated phone's display, the free phone opens (as individually set). If the free phone is already open, it is now closed.

- Symbols

User Interface Modules > easyCom > Symbols

There the symbols in the communication circuit and the image of the easyCom - design can be defined.

- Image

User Interface Modules > easyCom > Image

If you want the selected image to be represented in colors that match the current color scheme, enable the option Apply color scheme when drawing own images. If this option is disabled, the image will appear in optiClient 130 in its original state.

- Settings for SIP Functional Provider and Stimulus Provider in the Provider Module

Provider Modules > SIP Functional Provider

Provider Modules > Settings > Stimulus Provider

- Registrar and Proxy Settings:

Provider Modules > SIP Functional Provider > Registrar

Server: IP address of the HG 3500 board of **AMO CGWB**, parameter **IPADR**.

Provider Modules > SIP Functional Provider > Proxy

Server: IP address of the HG 3500 board of **AMO CGWB**, parameter **IPADR**.

- Settings of the Network access

Provider Modules > SIP Functional Provider > Network access

- Codec settings

Provider modules > SIP Functional Provider > Bandwidth Settings

- Settings of the used ports

Provider Modules > SIP Functional Provider > Port restrictions

- Quality of Service settings

Provider modules > SIP Functional Provider > Quality of Service

- Licensing

HiPath Provider > SIP Functional Provider > Licensing

Under **Server**, enter the IP address of the PC on which the license agent (CLA) for optiClient 130 is installed. For a local installation of the license server (as in the illustrated example), enter the local IP address or the address of the local host. If the license server is installed on the network, enter its IP address on the network.

- Configuration of the display control in Stimulus Provider

Provider Modules > Stimulus Provider > Display

- Key assignments

Manager Modules - Keyboard Manager - Key assignments

IMPORTANT: Only the function keys F2 up to F11 of the PC can be used!
Attention: You will delete the Windows functions, if you use the function keys for the optiClient V5.0!

9.6.2 Generating of the optiClient

1. Define the number of Common Gateway (HG 3500) boards:

```
ADD-DIMSU:TYPE=SYSTEM,CGW=xx;
```

2. Add the STMI board:

```
ADD-BFDAT:FCTBLK=1,FUNCTION=SIP,BRDBCHL=BCHL60;
CHANGE-BFDAT:CONFIG=CONT,FCTBLK=1,FUNCTION=SIP,LINECNT=10;
CHNAGE-BFDAT:CONFIG=OK,FCTBLK=1,ANSW=YES;

ADD-BCSU:MTYPE=IPGW,LTG=1,LTU=1,SLOT=49,PARTNO="Q2316-X",
FCTID=1,LWVAR="0",FCTBLK=1,BCHLSIP=10,ALARMNO=0;

CHANGE-BCSU:TYPE=HWYBDL,LTU=1,SLOT=49,PARTNO=Q2316-X,HWYBDL=A;

ADD-
CGWB:LTM=1,SLOT=49,SMODE=NORMAL,IPADR=1.40.41.105,NETMASK=255.255.255.0,DEFRT=1.40.41.254;
```

The Parameter **GWAUTREQ** and **SIPREG** must be set to NO,in situations where the board is used for SIP Subscriber.

```
CHANGE-CGWB:MTYPE=CGW,TYPE=SIPTRERH,GWAUTREQ=NO;
CHANGE-CGWB:MTYPE=CGW,TYPE=SIPTRSSA,SIPREG=NO;
```

3. Add optiClient 4222:

IMPORTANT: The parameter **AUTHREQ** must be set to **YES** and the security data has to be entered in the SIP software, if the parameters **SECZONE**, **USERID** and **PASSWD** are used in AMO SBCSU for a secured registration !

Add the primary station number or MSN (primary station number already exists):

e.g.:

```
ADD-SBCSU:STNO=4422,OPT=FPP,CONN=SIP,PEN=1-1-49-0,  
DVCFIG=S2PP,COS1=131,COS2=131,LCOSV1=9,LCOSV2=9,LCOSD1=1,  
LCOSD2=1,  
PROT="SBDSS1",OPTIDX=10,IPADR=1.40.41.2,PASSWD="SECRET",  
USERID="USER1",SECZONE="HIPATH4000.COM",FIXEDIP=YES,AUTHREQ=YES,  
SMGSUB=NO;
```

or e.g.

```
ADD-SBCSU:STNO=4422,OPT=MSN,MAINO=4420,COS1=131,COS2=131,  
LCOSV1=9,LCOSV2=9,LCOSD1=1,LCOSD2=1,FIXEDIP=NO,AUTHREQ=NO,  
SMGSUB=NO;
```

4. Activate the feature Direct Media Connection (DMC) for the station:

```
CHANGE-SDAT:STNO=4422,TYPE=ATTRIBUTE,AATTR=DMCALLWD;
```

9.7 Display of the SIP Subscribers

Registered and unregistered SIP subscribers (optiPoint or optiClient) can be displayed via the **HG 35xx Web Based Management** in the HiPath 4000 Assistant V5.

In the HiPath 4000 Assistant:

Expert mode -> HG 35xx Web Based Management

In the HG35xx Web Based Management:

Explorer -> Voice Gateway

Registered subscriber

Explorer -> Voice Gateway - Clients - SIP

Client Registered: Yes

Unregistered subscriber

Explorer -> Voice Gateway - Clients - SIP

Client Registered: No

9.8 Signalling of SIP Subscriber Outage

Requirements:

SIP subscriber endpoints are used in several situations at the customer site. Sometimes it is important to get informed, if a SIP endpoint gets out of service. This can be done towards a SNMP-System that receives, interprets and displays the content of SNMP-Traps.

Realization:

If SIP Phone goes out of service, SNMP Trap "MSC_ERH_SUB_OUT_OF_SERVICE" from IGW (HG3540/50) to the LAN network is reported. This SNMP trap includes information about time/date, IP address and station number.

User interface:

The reported TRAP can be verified via WMB access to the HG 3540 board:

Hipath4000 Assistant > Expert Mode > HG35xx WBM > Maintenance > SNMP > Traps (MSC_ERH_SUB_OUT_OF_SERVICE)

SIP Subscriber

Signalling of SIP Subscriber Outage

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