Residential and Centrex Services

IMT

Description

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Contents

1 Introduction

This document describes the rich set of Residential and Centrex services– traditional supplementary services, IP Centrex services, and other multimedia services provided by the system. The traditional services that the end-user may already be familiar with from the POTS world are enhanced and extended through the advantages of SIP.

1.1 Scope

The Residential and Centrex services address a complete list of services supported (except presence and VPN, described elsewhere) available to the Residential and Enterprise users. The functionality of the service is described, as well as the service interactions with other Residential and Centrex services.

This document describes the basic functionality of each service and provides an explanation of how the service is used provisioned by the user as well as an administrator. In addition, interactions between services are described whenever applicable.

Provisioning done by the Operator is described in the Provisioning Specification. See Reference 3.

The Self Provisioning capabilities are described in the Provisioning Specification. See Reference 3.

1.2 Target Groups

The target group is anyone who wants to learn more about different aspects of the product. Potential readers of the Function Specification may include

- Customer
- Product management
- Marketing specialists
- System developers

2 Survey of included Functions

2.1 Overview

User services – These services are assigned to specific users on the system and are used, managed, and configured by user.

Group services – These services apply to groups of users, which can be further categorized as:

> Virtual services – These services are assigned to a group and make use of a virtual user that performs some action upon receiving a call (for example, an Auto Attendant).
>
> Multi-user services – These services are assigned to a group and enable functionality that involves selected users in the group (for example, the Call Pickup service).
>
> Group services – These services provide functionality that applies to all users in a group (for example, an Outgoing Call Plan).

Messaging Services – These services provide users with the ability to send, receive and manage services.

Service provider and enterprise services – These services provide capabilities specific to the service provider administrator and the enterprise administrator.

Regular post paid charging of calls applies; no charging of individual services is used. The service invoking party is charged at the set up of a new call leg at the following services: Auto Attendant, Auto Callback, Call Forwarding, Call Intercept, Call Pickup, Call Transfer, Conference, Consultation, Directed Call Pickup, Directed Call Pickup with Barge-in, Hunt Groups, Simultaneous Ring Personal, Send VM Access Feature Code, Three Way Calling, Remote Office Origination and Third Party Voice Mail Support.

This document describes the basic functionality of each service and provides an explanation of how the service is used and configured by a user, as well as an administrator.

Centrex Service Application Server configuration and provisioning tasks are hierarchical and, although not explicitly mentioned for each occurrence, any configuration or provisioning action available to a user via a Commpilot can also be performed by administrators, such as a department administrator, group administrator or system provider. The same principle applies at each level.

The Services are described in alphabetic order.

Note: For PC-clients and SIP phones, switch hook and flash hook are not relevant. Instead, use appropriate buttons, e.g. buttons for hold, new call and resume.

3 Detailed Description

3.1 End-User Services, Originating Call Leg

3.1.1 Auto Callback

The Automatic Callback (ACB) service allows users to monitor a busy party and automatically establish a call when the busy party becomes idle.

Upon reaching a valid ACB busy condition, the user hears an announcement asking if they (he/she) would like to monitor the line and be called back when it is idle. To activate ACB the subscriber enters the digit prompted for then goes on hook. As soon as the called party becomes idle again, ACB attempts to re-establish the call between the subscriber and the previous busy party.

The ACB service can only be activated against a destination within the same group.

ACB is authorized and provisioned as a subscriber service, and can be enabled and disabled by the subscriber.

Description

Automatic Callback is an outgoing call feature that allows a subscriber to place a call to another subscriber in the same group. If the called subscriber is busy, the call originator can activate Automatic Callback to be notified when the called subscriber is idle. When notified, a new call setup attempt to the idle subscriber is initiated automatically and the originating subscriber is not required to redial the phone number. The new call attempt is treated as an originating call attempt; it could receive busy treatment or be redirected. For the new call setup to be attempted both parties must be available.

A terminating subscriber is considered busy, or unavailable if they cannot receive a call at their primary location. This means that if a terminating feature redirects the call and the new location is busy, ACB is not activated. ACB is also disabled if the call is handled by any of the following terminating services, Selective Call Rejection and Selective Call Acceptance, but is not limited to these service interactions.

When a subscriber originates a call to another subscriber in the group, if the called party is unable to receive the call because of a valid ACB busy condition, a prompt is played giving the originator the opportunity to activate ACB (for example, “The line you are calling is busy. Press 1 if you would like to be notified when the line is available”).

After activating ACB, the subscriber goes on hook and is notified with special ringing when both parties are idle. If the user answers special ringing, call setup is automatically initiated toward the other party.

A subscriber that has activated ACB can also deactivate all outstanding ACB requests. Entering the ACB deactivation Feature Access Code cancels all outstanding requests by that subscriber.

ACB has several operating parameters that are configured at the system:

- **Monitor Minutes**: The amount of time to camp-out on the busy line, waiting for it to become idle. (Default 30 minutes – Range 5 - 120)

- **Retry Originator Minutes**: The amount of time to wait before re-trying a busy Automatic Callback originator (the owner of the ACB session). (Default 5 minutes – Range 1 - 15)

- **Max Sessions**: The total number of outstanding ACB sessions for one subscriber. (Default 5 – Range 1 - 30)

- **Max Retry Rings**: The maximum number of rings when alerting the originator that the terminator is available. (Default 6 – Range 3 - 8)

- **Max Time To Retry**: Total amount of time to alert an originator that has been busy (unavailable). How long ACB tries to alert a busy originator. (Default 180 – Range 180 - 360)

Provisioning

At the system provider level, the administrator can modify the default Feature Access Code used to deactivate outstanding Automatic Callback sessions. This allows the service provider to set the deactivation code to match their legacy system. There is no requirement to change the Feature Access Code; the current default is #8.

Automatic Callback is a user assignable feature, once it is assigned to the use(r) they (he/she) have the option to enable or disable the feature. Automatic Callback is an outgoing feature with its default set to off. From the users Options web page, select Outgoing Calls, and then select Automatic Callback to provision the feature.

The following are system wide parameters for all users that are assigned Automatic Callback. The system administrator provides these parameters using the command line interface.

- **monitorLineMinutes** – How long the busy line will be monitored, in minutes. (Default 30, Range 5 - 120)

- **retryCallOriginatorMinutes** – If the originator that activated ACB is busy, how long to wait before trying again, in minutes. (Default 5, Range 1 - 15)
- **maxSessions** – Total number of active ACB session for one user. (Default 5, Range 1 – 30)
- **maxTimeToRetry** – Total time to alert a busy ACB originator. (Default 180, Range 180 – 360)
- **maxRetryRings** – The maximum of number of rings when alerting an ACB originator. (Default 6, Range 3 – 8)

3.1.2 Call Forwarding Remote Access

This service allows a user to activate, deactivate, and program the Call Forwarding Always service through an interactive voice response interface from any phone.

Description

This service allows users to configure their Call Forwarding Always service from any phone.

Users access this service by calling their group CommPilot Voice Portal from any phone.

After authenticating themselves with a user ID and a passcode, they are presented with a menu of options that includes the ability to query, activate, deactivate, and program their Call Forward Always service. After configuring this service, users may hang up, or continue to navigate through other options of the CommPilot Voice Portal.

Provisioning

The ability to configure Call Forward Always through the Voice Portal is automatically available to all users who are subscribing to the Call Forwarding Always service.

When the redirection number is configured, and validation option is enabled on system level, it is validated against the users calling plan profile and may be rejected.

3.1.3 Call Return

This service enables a user to call the last party that called, whether or not the call was answered. To call back the last party that called, the user dials a recall feature access code. The system stores the number of the last party that called and attempts to connect the user to that party.

Description

Call Return allows the user to call the last party that called by dialling *69 (default) on the user’s device or through the CommPilot Call Manager. Call Return can be used for calling back answered and unanswered calls, as long as the calling number is available to the Centex Service Application Server. If the calling number is available, the last calling party is called as if the user dialed this number directly. If the calling number is not available, the user is played an error announcement. A call originated with Call Return is subject to all user services and restrictions.

Note however, that when a user tries to use Call Return on a call with the caller ID blocked, the user is played an error announcement.

The Call Return service can be used through the CommPilot Call Manager. To do so, the user simply enters the Call Return feature access code in the CommPilot Call Manager dial window and clicks the *Dial* button. This results in holding the current call and originating a call to the last calling number if the user was already active on a call or it rings back the user’s phone and originates the call to the last calling number when the user picks up the phone.

Provisioning

There are no Provisioning parameters.

3.1.4 Call Trace (Customer Originated Trace)

This service enables the recipient of an obscene, harassing, or threatening call to request that it be automatically traced by dialling a feature access code after the call.

Description

Users are able to trace an incoming call by dialling *57 (default) after the call is received. The call that is being traced is defined as the call that was last received by the user. It could either be an answered or a missed call. If neither the name nor the number of the caller is available to the Centrex Service Application Server, a trace cannot be sent.

After dialling a feature access code, the user is played an announcement followed by dial tone. If neither the caller’s name nor number is available to the Centrex Service Application Server, the user gets an error announcement and the trace is not sent. Otherwise, the user gets a confirmation announcement and the requested trace is sent in the form of an alarm to the system provider.

The alarm contains the following information:

- The phone number of the user who initiated the trace. When the user does not have a phone number, the group phone number and the extension of the user is provided instead. For intra-group calls, only the caller’s extension is used.
- The date and time the call was received.
- The identity (name and number) of the caller, if available.

Provisioning

There are no Provisioning parameters.

3.1.5 Call Transfer with Three-Way Consultation

This service enables a user to make a three-way call with the original caller and an add-on party before transferring the caller to the add-on party.

Description

To initiate call transfer with three-way consultation, the user presses the flash hook, receives a dial tone, and then dials the add-on party. When the call is answered, the user presses the flash hook and forms a three-way call with the add-on party and the original caller. To transfer, the user hangs up which transfers the original caller to the add-on party.

Users can also execute call transfer with three-way consultation via the CommPilot Call Manager. To do so, they enter the number of the dial party in the dial window of the CommPilot Call Manager and click the *Dial* button. Once the call to the add-on party is established, users can click the *Conference* button to join the three parties together and have a three-way consultation before hanging up or using the transfer button to connect the two other parties together.

The user can select to abort the transfer during the three-way consultation by flashing the switch-hook, which releases the add-on party. The same can be achieved by releasing the party on the CommPilot Call Manager. The flash method and the CommPilot Call Manager buttons can be used interchangeably during the session.

Provisioning

This service has no Provisioning parameters.

3.1.6 Call Transfer with Third-Party Consultation

This service enables a user to consult with an add-on party before transferring the caller to the add-on party.

Description

To initiate call transfer with consultation, the user presses the flash hook, receives a dial tone, and then dials the add-on party. When the call is answered, the user can consult with the add-on party. To transfer, the user hangs up which transfers the original caller to the add-on party.

Users can also execute call transfer with consultation via the CommPilot Call Manager. To do so, they enter the number of the dial party in the dial window of the CommPilot Call Manager and click the *Dial* button. Once the call to the add-on party is established, users can talk privately with the add-on party before hanging up or using the *Transfer* button to connect the two other parties together.

The user can select to abort the transfer during the consultation by flashing the switch-hook twice, which releases the add-on party. The same can be achieved by releasing the party on the CommPilot Call Manager.

The flash method and the CommPilot Call Manager buttons can be used interchangeably during the session.

Provisioning

This service has no Provisioning parameters.

3.1.7 Calling Line ID Blocking Persistent

This service enables a user to persistently block delivery of their identity to the called party.

Description

Calling Line ID Blocking is used to block or allow the delivery of a user’s identity (both name and number) to a called party.

When active, calls made by the user to parties outside of the group have the presentation of their identity (name and number) blocked. The blocking is achieved by setting the presentation indicator associated with the calling party number to “private”, which prevents the user’s identity from being presented to the called party’s device.

The CLID Blocking flag can also be enabled via the user's phone by dialing the feature access code (FAC) *31#, and disabled by dialing the FAC #31# (as with all FACs, the exact digit sequence is configurable).

After the FAC is dialed, the system plays a confirmation announcement "Your Calling Line ID Blocking service is now set to [show | hide] your identity. Thank you". It then releases the call.

Provisioning

The user configures the service activation and deactivation via the CommPilot Personal web portal or via FAC.

3.1.8 Calling Line ID Blocking Per Call

This service overrides the persistent blocking of the calling line ID (CLID) so users can block the delivery of their identity for the next call. At the end of the call, the presentation of the user's identity is restored to its persistent status.

Description

The user blocks the delivery of their identity for the next call by dialling *67 (default), from their device, prior to making the call. This results in a confirmation tone followed by a dial tone. The user can then make the call as usual and their identity is blocked.

Per-call Calling Line ID Blocking is deactivated automatically when the user hangs up.

Feature access codes can also be used from the CommPilot Call Manager. For example, the user can enter *67XXXXXXXXXX to activate Per-call Calling Line ID Blocking with XXXXXXXXXX being the destination number. The system then rings the user's device and upon answer, starts alerting the called party destination. In addition, the user can just enter *67 through the CommPilot Call Manager. Then, the user's device rings, and when answered, the system prompts for the destination number (as described earlier).

If the user's identity is persistently blocked, the service is used the same way but has no impact.

Provisioning

This service has no Provisioning parameters.

3.1.9 Cancel Call Waiting Persistent

This service allows users to disable call waiting persistently.

Description

This service allows users to disable call waiting. When Cancel Call Waiting Persistent is active, any incoming call received while the user is already busy on a call receives busy processing. It is however possible for the user to have more than one call active if the user has originated them.

Provisioning

Persistent Cancel Call Waiting can be activated or deactivated through the CommPilot Personal web portal or by Feature Access Code (default #43).

3.1.10 Cancel Call Waiting Per Call

This service allows the users to disable Call Waiting for the next or current call.

Description

To cancel Call Waiting Per Call, the user dials *70 (default). The system plays a confirmation announcement and then applies a dial tone. The user then dials the destination number. For the duration of the call, the user is not presented with any waiting calls. Call Waiting is automatically re-activated when the call ends.

The user can also cancel Call Waiting for calls in progress. The user can flash the switchhook while a call is in progress, and then dial *70 (default) after the dial tone is applied. The system then responds with a confirmation announcement, followed by a dial tone. The user can then flash back to the other call, and no other waiting calls are presented for the duration of the current call.

Feature access codes can also be used from the CommPilot Call Manager. For example, the user can enter *70XXXXXXXXXX to cancel call waiting, with XXXXXXXXXX being the destination number. The system does not play a confirmation announcement. In addition, the user can just enter *70 through the CommPilot Call Manager. Then, the user's device rings, and when answered, the system prompts for the destination number (as described above).

Provisioning

Cancel Call Waiting Per Call has no Provisioning parameters and is available to all users with call waiting capabilities.

3.1.11 Consultation Hold

This service enables a user to put a caller on hold and then make a consultation call to another party.

Description

To initiate consultation hold, a user presses the flash hook and then dials the add-on party. When the call is answered, the user can consult with the add-on party. To drop the add-on party and reconnect to the original party, the user presses the flash hook twice.

Users can also execute Consultation Hold from the CommPilot Call Manager.

Provisioning

There are no Provisioning parameters.

3.1.12 External Emergency Routing

This service enhances the Centrex Service Application Server emergency calling support by optionally allowing the Centrex Service Application Server to query an Emergency Number Server to obtain a routable address to use for the emergency call.

Typically, emergency calls are dialled via a short number such as 911 or 112. The Emergency Number Server translates the emergency call and provides a routable address to the appropriate emergency service center based on the calling subscriber location.

The external emergency number database is not part of the IMT system.

Description

Activation of this service causes the Application Server to launch a query to the Emergency Number Server to obtain a routable number for the emergency number. This routable number is assumed to be a fully routable E-164 number to a suitable emergency call centre.

This service is enabled at the system level. The service has the address of the Emergency Number Server configured as an URI address. The Emergency Number Server is configured at the system level. Thus, the Application Server accesses the Emergency Number Server cluster via a DNS/DNS SRV query and selects a specific Emergency Number server.

If there is no reply from the Emergency Number Server, the Application Server falls back to default routing for the emergency call. A default emergency routing number is optionally configured on a system-wide basis. In the absence of a routing number obtained from the Emergency Number Server, the emergency call is routed to the default emergency number.

Provisioning

This service has no user Provisioning option.

3.1.13 Extension Dialling

This service enables users to dial extensions to call other members of their business group.

Description

Extension dialling allows a user to dial an abbreviated digit string to call another user in their group. By default, the extension is the last *n* digits of the user's phone number.

Extensions can be associated to users and virtual users.

Extensions can be dialled from the phone, from the CommPilot Call Manager dial button, or clicked from the CommPilot Call Manager group directory.

Callers to the group auto attendant can use the *dial by extension* option to reach any user of the group through their extension.

Provisioning

There are no Provisioning parameters.

3.1.14 Flash Call Hold

The Flash Call Hold feature allows a user to hold one call for any length of time until either party goes on-hook.

Description

When a subscriber is on a two-port call and wants to place the call on hold, the user flashes the switch-hook and receives a new dial tone.

Flash Call Hold is deactivated when any of the following occurs:

- If the holding party goes on-hook while another call is held, hold recall applies and the holding party is alerted with power ringing. When going off-hook, the connection with the held party is restored.

- The held station hangs up, terminating the connection.

- The holding station flashes the switch-hook and is then reconnected to the held party.

This service has specific interactions with other services, e.g. Call Waiting and Consultation Hold.

Provisioning

This service has no user Provisioning option.

3.1.15 Last Number Redial

This service enables users to redial the last number they called by clicking the Redial button on their CommPilot Call Manager or by dialling a feature access code (for example, *66).

Description

The Last Number Redial service allows the user to repeat the last call that was made by dialling *66 (default). The service substitutes the feature access code with the digits used for the last call that was made and originates the call as usual. The digits are obtained from the last entry in the call log for originated calls.

Feature access codes can be entered on the user's device or through the CommPilot Call Manager. Furthermore, the service can be used on the original or add-on call leg (that is, after a switch-hook flash or through the CommPilot Call Manager with one call active).

Provisioning

There are no Provisioning parameters.

3.1.16 Physical Location

This service controls whether originating calls are allowed from physical locations other than the physical location configured for the user. The service can be turned on and off for a user by the system administrator.

Description

Physical Location allows the operator to filter out calls where the information about the calls origin is insecure or missing. The intention with the service is primarily to allow proper support of emergency calls in countries and regions where the location of a user is insecure or missing.

When a user has the Physical Location service assigned and set to “Off”, it has no effect on any calls.

When a user has the Physical Location service assigned and set to “On”, the user’s physical location is compared with the configured physical location for the originating device.”

- If the user’s physical location is equal to the configured physical location, then the call is allowed to proceed.

- If the user’s physical location is not equal to the configured physical location, then the call is blocked and a treatment is played. Note that all originating calls (including emergency calls and repair calls) are blocked.

Provisioning

When Physical Location is assigned to a user, the system administrator specifies if the service is activated or not.

The configured Physical Location is a string value of 1024 bits representing the physical location of the user. This value is read-only to the user and read/write to group, service provider, enterprise and system administrators.

3.1.17 Speed Dial 8

This service allows users to associate single-digit codes to frequently dialled or hard-to-remember long strings of digits. Users can then use these codes instead of the full numbers to place calls.

Description

A user can associate a single-digit code with a string of digits. This single-digit code is referred to as a speed code. The user can associate two to 30 numeric digits, including emergency and repair call numbers, or a URL. The user can also associate a string of digits that includes the * character. This allows a user to program a speed code that maps to a feature access code, or multiple, chained feature access codes.

At the time of programming, the system does not validate the digits. It is the responsibility of the user to ensure that the digit string associated with a speed code is valid. When the associated digit sting is invalid, the user is provided with the applicable treatment.

Once a speed code is defined, the user can dial that single digit and wait (for an inter-digit timer to expire), or terminate the code with a #. This can be done through the user's device, or from the CommPilot Call Manager. Once the speed code is collected, it is replaced by its associated digit string and the call is originated as usual with these digits.

Provisioning

There are two methods by which a user can program speed codes, through the CommPilot Personal web portal or by dialling a feature access code.

CommPilot Personal Web Portal

A table of eight rows contains the speed code definitions. A user can enter the digits or URL and a name or description for the speed code.

Feature Access Code

The user can program speed codes by dialling *74 (default) from their device. The user is played recall dial tone (three quick beeps and then regular dial tone). The user then dials the single-digit speed code, followed by the associated phone number and a terminating digit (#). The system plays a confirmation announcement and the user hangs up.

Feature access codes can also be used from the CommPilot Call Manager. For example, the user can enter *74YXXXXXXXXXX to define a speed code, where Y is the speed code and XXXXXXXXXX is the associated phone number. The system then rings the user's device and plays a confirmation announcement indicating that the speed code has been defined. In addition, the user can just enter *74 through the CommPilot Call Manager. Then, the user's device rings, and when answered, the system prompts for the speed code and associated phone number (as described above).

3.1.18 Speed Dial 100

Description

The Speed Dial 100 service allows users to place calls using a directory of up to 100 frequently called numbers. The user dials the associated two-digit speed code, which is preceded with a configurable prefix. This is a user-assignable service that involves two activities: programming and dialling.

Programming

A user can associate a Speed Dial 100 prefix and two-digit code with a string of digits or a URL. The Speed Dial 100 prefix can be configured from the Feature Access Code page and the code can be one or two digits selected from the following: [0-9, A-D, *, #]. (The default value for Speed Dial 100 prefix is the # character.)

The user can associate 2 to 30 numeric digits, including emergency and repair call numbers or a URL. The user can also associate a string of digits that includes the * character. This allows a user to program a speed code that maps to a feature access code, or multiple chained feature access codes.

Apart from the criteria listed above, the Speed Dial 100 service performs minimal validation on the digit string associated with a speed code. It is the responsibility of the user to ensure that the digit string associated with a speed code is valid. If the associated digit sting is invalid, the user is provided with the applicable treatment.

There are two methods by which a user can program speed codes, through the CommPilot web portal or through a phone using feature access codes.

- CommPilot Web Portal

When the user selects the “Speed Call 100” option from the Outgoing Calls menu, a list appears that displays the personal speed codes that are currently configured.

To configure a new Speed Dial 100 speed code, a user clicks the Add button on the *Speed Dial 100* Provisioning page. For each speed code, the user enters the code and the associated digit string or URL, as well as an optional short description of the code (for example, Company’s Auto Attendant).

To modify an existing speed code, a user clicks the “Edit” link on the Speed Call 100 page, which redirects the user to the *Speed Call 100 Modify* page. This page is similar to the “Add” page.

- Feature Access Code

The user can program Speed Dial 100 speed codes through a phone using a feature access code. The phone can be an analogue phone, an IP phone, or even the CommPilot Call Manager (with any type of phone).

To program through the phone, the user dials the Speed Dial 100 feature access code, (default is *75). The user then hears the “recall dial tone” (three quick beeps and then a dial-tone). The user then dials the two-digit speed code to be created or modified, followed by the digits to be associated with it. The user then ends the programming with # or waits for the inter-digit timer to expire. The service then announces the success or failure of the programming.

For example, if a user wants to program the speed code 23 with the following digits 0112511792402, the user enters:

*75 [recall dial tone] 23 0112511792402# [success or failure announcement]

Any change to a user's Speed Dial 100 speed codes using a phone are reflected on the user's *Speed Dial 100* Provisioning page.

Dialling

Once a speed code is defined, the user can dial that speed code and end dialling with the # or wait (for an inter-digit timer to expire). This can be done through the user's phone or the CommPilot Call Manager. However, there are several interactions that can interfere with Speed Dial 100 dialling:

- Collision between a feature access code and Speed Dial 100
- Collision with an emergency number or repair number
- Collision with an extension number

Collision Between a Feature Access Code and Speed Dial 100

This scenario occurs when a speed code uses the same prefix and digits as an existing feature access code. For example, if the user creates the *70 speed code, this may collide with the default *70 Cancel Call Waiting feature access code. In this case, the feature access code takes precedence over the speed code.

Collision with an Emergency Number or Repair Number

This scenario occurs when a speed code plus its prefix is the same as an emergency or repair number.

Collision with an Extension Number

This scenario occurs when a Speed Dial 100 prefix and speed code are the same as an extension number. For example, suppose a group has its extension dialling set to four digits and a user in that group uses the default feature access code for Speed Dial 100 (*75) and programs 0112511792402 as follows: *75 52 29 0112511792402. In this scenario 52 is the prefix for Speed Dial 100 and 29 is the two-digit speed code to call 0112511792402. If another user in that group has its extension number set to 5229, in this situation dialling 5229 by the first user would cause the extension number not the speed code to be dialled. In this case, an extension number take precedence over Speed Dial 100.

<u>CommPilot Interaction</u>

The user can also use Feature Access Codes through the CommPilot Call Manager. In addition to the interactive sequence described above, the user for example can enter *75 ## XX 0112511792402# to program XX as the speed code for 0112511792402. Where ## is the Speed Dial 100 prefix. The user's phone then rings, the user answers, and then hears an announcement indicating the programming has succeeded or failed.

Provisioning

For a description of the user Provisioning of Speed Dial 100 codes, see section *Programming.*

Speed Dial 100 Feature Access Code Assignment Provisioning

A group administrator and/or a higher-level administrator can provision the Feature Access Code that is used to program Speed Dial 100 for a given group. The system default is *75, but a group administrator and/or a higher-level administrator can set the Feature Access Code to an arbitrary value through the web interface. The Feature Access Code can be provisioned once Speed Dial 100 is authorized for a group. At the time Speed Dial 100 is authorized, if another feature already is using *75 Feature Access Code, the Speed Dial 100 is set to "blank". The Feature Access Code for Speed Dial 100 cannot be used until a new code is selected.

3.1.19 Three-Way Calling

This service enables a user to make a three-way call with two other parties, whereby all parties can communicate with each other.

Description

Analogue Phone

To initiate a three-way call while engaged in a regular two party call, the user presses the flash hook and dials the third party (also known as an add-on party). Before or after the add-on party answers, the user presses the flash hook and forms a three-way call with the two parties. To drop the add-on party, the user presses the flash hook and is reconnected with the original party in a regular two party call. If the user hangs up, all parties are released (unless the user is allowed to transfer, in which case the call remains active between the two other parties).

The user can have a private conversation with the add-on party before conferencing all three parties. This phase is known as consultation. If the user hangs up during consultation, the add-on party is released and the user is recalled by the original party on hold (that is, the user's device is rung again).

CommPilot Call Manager

Users also have the ability to execute three-way calls using the CommPilot Call Manager. To initiate a conference, the user simply clicks the conference button on the CommPilot Call Manager while two calls are active. From this point, all three parties get connected on a three-way call.

The CommPilot Call Manager also provides the user with capabilities above and beyond what can be done on an analogue phone, as follows:

- *Conference hold* - A user can select the conference and hold it, which results in muting the user from the conference.

- *Conference release* – A user can select the conference and release it, which results in dropping all parties, independently of the call transfer capability.

- *Call waiting/join* – A user involved in a call waiting session has no way to join all parties in a conference using an analogue phone. The flash is always interpreted as a toggle between the two active parties. The CommPilot Call Manager allows users to join parties involved in a call waiting session with the conference button.

- *Selective hold* – A user can hold either active party at any time, as many times as desired, with the Hold and Talk (retrieve) buttons.

- *Selective release* – A user can selectively release any party, not just the add-on parties.

IP Phones

IP phones have access to the same capabilities as analogue phones through the CommPilot Call Manager, and often provide embedded conferencing capabilities on the phone itself. An IP phone with embedded conferencing capabilities and corresponding procedures are specific to each phone and vendor.

Provisioning

This service is provided by default to users assigned the CommPilot Call Manager web control client. IP phone users can also establish three-way conferences through the phone's feature keys.

3.2 End-User Services, Terminating Call Leg

3.2.1 Anonymous Call Rejection

This service enables a user to reject calls from anonymous parties who have explicitly restricted their identity. By activating the service, callers who have restricted their identity are informed that the user is not accepting calls from restricted callers. The user's phone does not ring and the user does not see or hear any indication of the attempted call. This service does not apply to calls from within a group.

This service does not apply when the calling party's identity is not available, only when the calling party has explicitly restricted their identity.

<u>Description</u>

Anonymous Call Rejection enables users to instruct the Centrex Service Application Server to reject incoming call attempts from callers not within the same group, who have blocked their identity (phone number and name) to the user, with a calling identity delivery blocking service.

When this service is active, the user receives no alerting indication for external calls from callers with their identity blocked. Instead, the caller is connected to an announcement stating that the user does not accept calls with the caller's identity blocked.

Another common name for this service is Anonymous Caller Rejection but the Anonymous Call Rejection naming convention is used throughout the documentation and system for consistency.

<u>Provisioning</u>

The user configures this service through the CommPilot Personal web portal. The service can be activated (block anonymous calls) and deactivated (allow anonymous calls).

3.2.2 Automatic Hold/Retrieve

The Automatic Hold/Retrieve (AHR) service provides an alternate method to hold and retrieve calls for Centrex Application Server users. This service is assigned to users so that their incoming calls are automatically held and retrieved without having to use feature access codes.

This service is useful for attendants who handle many incoming calls, by allowing them to hold calls simply by transferring them to dedicated parking stations. This service also allows the holding of calls without having to use a flash key, which many SIP CPEs do not provide.

Description

The AHR service is commonly used by receptionists operating attendant consoles. When a call terminates on the attendant console, the receptionist answers the call, gathers information from the caller, and then transfers the call to a subscriber with the AHR service. The caller is held and listens to Music On Hold while waiting. The receptionist then communicates with the person who should handle the call, and provides him/her with the extension against which the call is held. That person calls the extension and retrieves the call that is on hold.

Only one call can be held for a subscriber with the AHR feature active.

Provisioning

This service is authorized at the service provider and group levels and is assigned at the user level. It introduces two new attributes for a user:

Active – The AHR service is enabled when this attribute is set to “true” and disabled when this attribute is set to “false”.

Recall Timer – The AHR recall functionality is activated after a call has been held for the value specified for this attribute.

These attributes can be set by end users, group administrators, and administrators using the web portal, or through the OSS interface.

3.2.3 Blind Call Transfer

This service enables a user to transfer a call before or after the call is answered, without consulting with the transferred to party. Users can only execute blind call transfer from the CommPilot Call Manager.

Description

Blind Call Transfer allows a user to transfer an active call to a specific destination without consulting with the destination party. This capability is provided exclusively through the CommPilot Call Manager.

To blind transfer an incoming call (unanswered), the user selects the incoming call in the active call window, enters a destination in the dial window, and clicks the Transfer button. The call is automatically redirected to the specified destination.

To blind transfer a talking or held call, the user selects the call in the active call window, enters a destination in the dial window, and clicks the Transfer button. The call is automatically redirected to the specified destination. When there is only one call in the active call window, this call is selected by default and does not need to be selected explicitly.

Provisioning

There are no Provisioning parameters. The Blind Call Transfer service is provided as part of the CommPilot Call Manager.

3.2.4 Call Forwarding Always

This service enables a user to automatically redirect incoming calls to another destination.

Description

Call Forwarding Always provides the capability to redirect incoming calls intended for a user to another destination. When active, incoming calls are redirected (that is, busy, idle, alerting, and so on) [1].

When the service is active, a reminder indicator is set on the user's CommPilot Call Manager. Furthermore, a ring splash (500 ms ring burst) is applied to the user's device each time a call is forwarded[2].

The Centrex Service Application Server supports multi-path forwarding for all flavours of call forwarding. Thus, there are no restrictions on the number of simultaneous forwarded calls.

The Outgoing Calling Plan service allows a group administrator to impose restrictions that are specific to calls forwarded from a user, therefore eliminating a fraud exposure that may result from uncontrolled forwarding of calls.

Provisioning

This service can be controlled via the CommPilot Personal web portal, via feature access codes dialled from the user's device and from the CommPilot Voice Portal. The first two are described below. The Provisioning through the CommPilot Voice Portal is described in *Section Call Forwarding Remote Access*.

When the redirection number is configured, and validation option is enabled on system level, it is validated against the users calling plan profile and may be rejected.

[1] For intra group calls, looping calls are detected and CFA will not apply, e.g. A & B are in same group, A calls B and B has CFA to A. The call will be set up.
[2] The ring burst capability is optional and can be turned on or off by the user

Web Portal Activation

The Call Forwarding Always service can be activated and deactivated through the user's CommPilot Personal web portal. When activated, a valid forwarding phone number or URL must be entered. Users can also select whether a ring splash should be played upon forwarding a call.

The Call Forwarding Always Provisioning page can also be accessed directly by clicking on the CFA button on the CommPilot Call Manager.

Feature Access Codes

The Call Forwarding Always service can be activated and deactivated through feature access codes dialled from the user's device.

To activate, the user dials *72 (default), optionally followed by a valid forwarding phone number. If no phone number is entered, the calls are forwarded to the phone number that was previously configured by default. The system then plays a confirmation announcement and the user hangs up.

To deactivate, the user dials *73 (default). The system then plays a confirmation announcement and the user hangs up.

Feature access codes can also be used from the CommPilot Call Manager. For example, the user can enter *72XXXXXXXXXX to activate call forwarding, with XXXXXXXXXX being the forwarding number. The system then rings the user's device and plays a confirmation announcement indicating that call forwarding has been activated.

In addition, the user can just enter *72 through the CommPilot Call Manager. Then, the user's device rings, and when answered, the system prompts for the forwarding number (as described earlier).

3.2.5 Call Forwarding Busy

This service enables a user to redirect incoming calls to another destination when the user is busy.

<u>Description</u>

Call Forward Busy forwards calls to a specified destination when the user is busy. A user is considered busy when there are too many calls active or a service makes the user appear busy to the caller (for example, the Do Not Disturb and Selective Call Rejection, services).

<u>Provisioning</u>

The service can be controlled via the CommPilot Personal web portal or via feature access codes dialled from the user's device.

When the redirection number is configured, and validation option is enabled on system level, it is validated against the users calling plan profile and may be rejected.

Web Portal Activation

The service can be activated and deactivated through the CommPilot Personal web portal. When activated, a valid forwarding phone number or URL must be entered.

Feature Access Codes Activation

The service can be activated and deactivated through feature access codes dialled from the user's device. To activate, the user dials *90 (default), optionally followed by a valid forwarding phone number. The system then plays a confirmation announcement and the user hangs up. To deactivate, the user dials *91 (default). The system then plays a confirmation announcement and the user hangs up. Feature access codes can also be used from the CommPilot Call Manager. For example, the user can enter *90XXXXXXXXXX to activate call forwarding with XXXXXXXXXX being the forwarding number. The system then rings the user's device and plays a confirmation announcement indicating that call forwarding has been activated. In addition, the user can just enter *90 through the CommPilot Call Manager. Then, the user's device rings, and when answered, the system prompts for the forwarding number (as described above).

3.2.6 Call Forwarding No Answer

This service enables a user to redirect incoming calls to another destination when the user does not answer within a specified number of rings.

<u>Description</u>

Call Forwarding No Answer forwards calls to a specified forwarding phone number when a user does not answer an incoming call for a user specified number of rings.

<u>Provisioning</u>

The service can be controlled via the CommPilot Personal web portal or via feature access codes dialled from the user's device.

When the redirection number is configured, and validation option is enabled on system level, it is validated against the users calling plan profile and may be rejected.

Web Activation

The service can be activated and deactivated through the user's CommPilot Personal web portal. When activated, a valid forwarding phone number or URL must be entered. Users can also configure the number of rings before the call is forwarded.

Feature Access Codes Activation

The service can be activated and deactivated through feature access codes dialled from the user's device. To activate, the user dials *92 (default), optionally followed by a valid forwarding phone number. The system then plays a confirmation announcement and the user hangs up. To deactivate, the user dials *93 (default). The system then plays a confirmation announcement and the user hangs up.

Feature access codes can also be used from the CommPilot Call Manager. For example, the user can enter *92XXXXXXXXXX to activate call forwarding with XXXXXXXXXX being the forwarding number. The system then rings the user's device and plays a confirmation announcement indicating that call forwarding has been activated. In addition, the user can just enter *92 through the CommPilot Call Manager. Then, the user's device rings, and when answered, the system prompts for the forwarding number (as described earlier).

It is possible for the user to change the number of rings required for No-Answer handling (same value as for other services like Sequential Rings) by dialing a feature access code (FAC) code from his/her phone.

1) From a plain phone:

- User dials *610.
- User listens to the prompt "Please enter the number of rings before the system applies No-Answer handling for your incoming calls".
- User punches one digit (valid digits are 0, 2, 3, 4, 5, 6).
- User hears the announcement "Thank you. The system will apply No-Answer handling after… 'n' …rings for your incoming calls".

2) From a SIP phone or call manager, the user can still use the FAC as stated above, or also:

- User dials *610*n#.
- User hears the announcement "Thank you. The system will apply No-Answer handling after… 'n' …rings for your incoming calls".

When the value of *number of rings required for No-Answer handling* is set to “0”, the system will generate a No-Answer condition before the Voice Mail service sees the incoming call. As a result, the No-Answer handling will apply.

3.2.7 Call Forwarding Selective

This service enables a user to define criteria that causes certain incoming calls to be redirected to a user specified destination.

Description

Call Forwarding Selective provides the capability to forward calls intended for a user to another destination when the incoming call matches pre-specified criteria. If the incoming call does not match any of the criteria, normal call handling applies.

The possible criteria include:

Selected time schedule, for example, “Every Day All Day”

Whether the calling line ID is PRIVATE or UNAVAILABLE

A list of up to 12 phone numbers or digit patterns (for example, 514*)

The criteria can be combined within predicates (for example, incoming call from this number AND within business hours AND during work week). Multiple predicates can be defined and the call is forwarded when at least one of the predicates is met.

The user can associate a different destination with each predicate, or use the same destination for all predicates.

Provisioning

The service is configured through the CommPilot Personal web portal. The user defines criteria based on the incoming caller identity, ranges of digits, time schedule, and the inter/intra-group status of the call.

When the redirection number is configured, and validation option is enabled on system level, it is validated against the users calling plan profile and may be rejected.

Ranges of digits can include digits from 0-9, and the following wildcard characters:

- * (star) – This wildcard can only be used as the last character of a digit string and matches any number of trailing digits.

- ? (question mark) – This wildcard can be used anywhere in the string and matches any single digit.

Multiple criteria can be combined to build predicates, and multiple predicates can be defined simultaneously. Each predicate can be active or inactive. Each predicate can be associated with its own destination number or URL.

3.2.8 Call Notify

This service enables a user to define criteria that cause certain incoming calls to trigger an e-mail notification to a user-specified address.

Description

When an incoming call matches pre-defined criteria, this service sends an e-mail with information about the caller to a user-configurable address. The criteria include:

Selected time schedule, for example, “Every Day All Day”

Whether the calling line ID is PRIVATE or UNAVAILABLE

A list of up to 12 phone numbers or digit patterns (for example, 514*)

The criteria can be combined within predicates (for example, incoming call from this number AND within business hours AND during work week). Multiple predicates can be defined and a notification is sent when at least one of the predicates is met.

Provisioning

This service is configured through the CommPilot Personal web portal. The user defines criteria based on the incoming caller identity, ranges of digits, the time of day, and the day of the week.

Ranges of digits can include digits from 0-9, and the following wildcard characters:

* (star) – This wildcard can only be used as the last character of the digits string and matches any number of trailing digits.

? (question mark) – This wildcard can be used anywhere in the string and matches any single digit.

Multiple criteria can be combined to build predicates, and multiple predicates can be defined simultaneously. Each predicate can be active or inactive. The user defines an e-mail address for notification, which is used to send the notification if any of the active predicates is met.

3.2.9 Call Waiting

This service enables a user to answer a call while already engaged in another call.

Description

When an incoming call is received while a user is already engaged in a call, the user is informed of the new call via a call waiting tone. To answer the waiting call, the user presses the flash hook, which connects the user with the waiting party and holds the original party. Subsequent use of the flash hook allows the user to toggle between the two parties.

If the user hangs up while another party in the session is held or waiting, the user is rerung.

Upon answer, the user is reconnected to the held party.

The service ends when any party hangs up.

Users can also execute call waiting via the CommPilot Call Manager. When a second call is presented to the user in the CommPilot Call Manager window, the user can click the *Talk* button while the new party is highlighted to hold the other party and establish a connection with the incoming call. This procedure can be repeated as many times as necessary to toggle between the two parties.

The flash method and the CommPilot Call Manager can be used interchangeably during the same session.

Call Waiting (persistent) setting can also be enabled via the user's phone (or Call Manager) by dialing the feature access code (FAC) *43#, and disabled by dialing the FAC #43# (as with all FACs, the exact digit sequence is configurable).

After the FAC is dialed, the system plays a confirmation announcement "Your Call Waiting service is [enabled | disabled]. Thank you". It then releases the call.

Provisioning

This service has no provisioning option.

A user cancels call waiting by pressing *70.

Call waiting can persistently be activated by pressing *43 and deactivated by pressing #43.

3.2.10 Calling Line ID Delivery

This service is a terminating service that delivers the identity of the calling party to the user via the CommPilot Call Manager and device (if capable).

Description

Calling Line ID Delivery relays a caller's identity to the user's CommPilot Call Manager and device, if the device is capable of displaying such information.

The caller identity is delivered for every call that terminates to the user. If an incoming call is redirected or blocked before it can terminate, or if the user is busy, the identity is not delivered. The identity includes the calling party's number and name, if available.

Provisioning

This service has no user Provisioning parameters.

3.2.11 Calling Line ID Delivery Per Call

This service overrides the persistent presentation of the calling line ID (CLID) so users can allow the delivery of their identity for the next call. At the end of the call, the presentation of the user's identity is restored to its persistent status.

Description

This service is the exact opposite of Calling Line ID Blocking Per Call (see *Calling Line ID Blocking Per Call*) and shares the same characteristics of the user interface and service interactions.

The user allows the delivery of their identity for the next call by dialling *65 (default), from their device, prior to making the call. This results in a confirmation tone[3] followed by dial tone. The user can then make the call as usual and their identity is delivered to the far end.

When the user hangs up, the blocking of the calling line ID is restored to its persistent status.

[3] This is the same type of announcement/tone as Calling Line ID Blocking Per Call.

Feature access codes can also be used from the CommPilot Call Manager. For example, the user can enter *65XXXXXXXXXX to activate Calling Line ID Delivery Per Call (the XXXXXXXXXX represents the destination number). The system then rings the user's device and upon answer, starts alerting the called party destination. In addition, the user can simply enter *65 through the CommPilot Call Manager. In this case, the user's device rings, and when answered, the system prompts for the destination number (as described above).

Calling Line ID Delivery Per Call behaves the same if the CLID is not persistently blocked for the user, but it has no impact.

<u>Provisioning</u>

This service has no Provisioning parameters.

3.2.12 Calling Line ID Restriction Override

Calling Line ID Blocking Override (CLIO) allows a user to override calling line identity presentation restrictions and always receive the calling line identity at their CPE, if available.

In other words, the user who activates this option always receives the calling line identity if available, regardless of whether or not the calling line identity is blocked. The user never receives a calling line identity indicating "private". This service is configured via the CommPilot web portal.

<u>Description</u>

Calling Line ID Blocking Override is offered as an override to Calling Line ID Delivery Blocking. When activated by the user, this service ignores the presentation indicator and delivers the calling line ID to the user if it is available (both name and number).

The user activates and deactivates the service persistently through the CommPilot Personal web interface.

It should be noted that the caller information provided to the CLIO user is bound to what Centrex Service Application Server receives from the calling party. If the service is activated and the incoming call is from abroad, from a payphone or from an operator, the CLIO user only gets the corresponding indication instead of the caller ID.

When this service is active, all Caller ID-based services behave as if the Caller ID was present, regardless of the presentation indicator (unless the call is ANONYMOUS):

- Anonymous Caller Rejection lets private calls through.
- Screening services apply regardless of the presentation indicator.

- The call manager incoming call log show all incoming Caller Ids.

This service is only overridden by the CLID Delivery (both External and Internal) service. If the Caller ID Delivery service is not active for a user, then CLIO has no impact.

Provisioning

The service is activated and deactivated via the user's web portal.

3.2.13 Calling Name Delivery

Description

This service allows BroadWorks to provide calling name delivery to its users by retrieving the calling name from a PSTN-hosted database.

If the name information is already present in the incoming call setup message, then the external database is not accessed.

BroadWorks uses the SIP event notification framework to communicate with the external database. A SIP Subscribe message is sent by BroadWorks to query the caller's name information, and the external database uses a SIP Notify message to respond to the BroadWorks request.

The query contains the caller's number, which allows the external database to query the caller's name information. When BroadWorks receives a response from the external database, the caller's name information is extracted from the message and is relayed to the called endpoint.

Provisioning

The service is authorized and assigned as usual to the user and can be activated/de-activated.

3.2.14 Custom Ringback User

Description

The two Custom Ringback user services (one for audio and one for video) allows a user to specify custom media files to be used for ringback, when incoming calls are received. When the user is called, the system allocates a media resource and plays a custom ringback file to the caller instead of the standard ringback tone.

The user services allow a user to specify multiple profiles. Each profile is associated with a set of criteria (phone numbers, time of day, and so on) and a custom media file. When a call is received, it is compared with the profiles associated with the user. If a match is found, then the associated custom media file is used; otherwise the group service is checked. If active, then the group's custom media file is used; otherwise system ringback is provided.

Provisioning

Custom Ringback User service allows a user to configure selective profiles, with each profile having its own defined audio Custom Ringback. Custom Ringback selective profiles conform to the Centrex Application Server model for selective services.

When the Custom Ringback User - Video service is assigned, it enhances the Custom Ringback service. It allows a video ringback to be configured in addition to an audio ringback in each Custom Ringback selective profile.

NOTE: For each Custom Ringback selective profile, at least one Custom Ringback file (audio or video) is defined.

3.2.15 Directed Call Pickup

Directed Call Pickup allows a user to dial a feature access code followed by an extension, to pick up (answer) a call directed to a user with that extension in the same customer group.

Description

To pick up a call using the Directed Call Pickup service, users dial the Directed Call Pickup feature access code, followed by the extension of the ringing party.

If the ringing party has already answered the call, or if it has no alerting call, or if the dialed extension is invalid, the user receives a reorder treatment.

The main feature interactions introduced by this service are described in the following table.

Feature	***Interaction Description***
Call Waiting	It is not possible to pick up a waiting call. A call must be alerting the user with ringing to be picked up.

Call Forwarding No Answer	It is possible to pick up a ringing call before the call is forwarded by Call Forward No Answer (CFNA). Picked-up calls are not forwarded by the user picking up the call.
Call Hold and Retrieve	It is possible to flash during a call to place a call on hold and pick up another call. In addition, the user can use the CommPilot to hold the active call and dial the Call Pickup access code.
Call Notify	Call Pickup does not send a call notify message for picked-up calls.
Call Transfer (Blind Transfer)	It is possible to pick up a blind transferred call. It is possible to transfer a picked-up call.
Caller ID	When the identity of the calling party is anonymous, the caller's identity is not delivered to the user who picks up the call.
Do Not Disturb	It is possible to pick up calls regardless of whether the answering party is accepting calls.
Three-Way Call	It is not possible for the conference controller to pick up another call. If the controller flashes during a conference, the flash is processed in priority by the flash service. However, a participant in a three-way call can pick up another call by flashing and dialing the call pickup access code.

<u>Provisioning</u>

This service has no user provisioning option.

The administrator configures the Directed Call Pickup feature access code on the group *Feature Access Code* page.

3.2.16 Directed Call Pickup with Barge-in

Directed Call Pickup with Barge-in (DPUBI) allows users to dial a feature access code (FAC) followed by an extension to pickup (answer) a call directed to another user in the same customer group, or barge-in on the call if the call was already answered. When a barge-in occurs, a three-way call is established between the parties with the DPUBI user as the controller.

NOTE: The pick up portion of this feature is identical to the existing Directed Call Pickup (DPU) feature. DPUBI is a completely separate service from DPU, however, that adds the barge-in capability and has its own, unique FAC.

Throughout this section, the following terms apply:

- DPUBI user – the user invoking the DPUBI service

- Picked up user – the user whose extension has been selected by the DPUBI user

- Other party – the party that is connected with the picked up user before the DPUBI attempt takes place.

Description – Provisioning

The DPUBI service is a user service. Therefore, it can be authorized to service providers and groups, and can be assigned to users.

The only configurable option for the DPUBI user service is the barge-in warning tone. This option controls whether a warning tone is given to the picked up user when a barge-in occurs.

The warning tone option is only configurable by administrators. Users can view the current warning tone setting, but cannot change it.

At the group level, administrators can configure the DPUBI FAC. It defaults to *33, but can be changed to any valid, available FAC.

Description – Service Invocation

A user invokes the DPUBI service by dialing the DPUBI FAC. If the user does not supply an extension when dialing the DPUBI FAC, then they are given stutter dial tone so that they can enter the extension to be picked up.

If an invalid extension is entered (for example, an extension that doesn't exist in the group, too few digits, etc.), then the DPUBI user is given reorder tone.

If a valid extension is entered, then a pickup or barge-in is attempted as described in the following sections.

NOTE: If the picked up user has no calls or more than one call, then the DPUBI user is given reorder tone. A pickup or barge-in can occur only when the picked up user has exactly one call.

Description – Pickup

A pickup is triggered by the DPUBI service when the picked up user is alerting for a single, terminating call. When a pickup occurs, the DPUBI user and the other party are connected to one another, and the picked up user is released.

The DPUBI service's pickup functionality is identical to the DPU service, so please refer to the Release 10 Directed Call Pickup FS for any additional information.

Description – Barge-in

A barge-in is triggered by the DPUBI service when the picked up user has a single, answered call. The barge-in occurs regardless of whether the picked up user's call was originating or terminating, and regardless of its current state (for example, active and held).

When a barge-in occurs and the user's warning tone option is enabled, the picked up user is given the barge-in warning tone (1 second of 440 Hz followed by 50 ms of silence). The other party is put on hold while the picked up user is receiving the warning tone.

NOTE: The picked up user is not given the warning tone if they have put the call on hold.

Once the warning tone has finished (or immediately if the user's warning tone option is disabled), a three-way call is established with the DPUBI user as the controller. The DPUBI user now has a call with the picked up user and the other party. The picked up user and the other party now have a call with the DPUBI user instead of with each other.

If the DPUBI user has the Flash Three-Way Call service assigned and flashes while the three-way call for the barge-in is present, then the other party is dropped from the three-way call. If the DPUBI user does not have the Flash Three-Way Call service and flashes, then the flash is ignored per the existing Flash service rules.

If the DPUBI user has the Flash Transfer service assigned and hangs up (goes on-hook) while the three-way call for the barge-in is present, then the picked up user and the other party are transferred together. If the DPUBI user does not have the Flash Transfer service assigned and hangs up, the picked up user and the other party are released.

Description – Barge-in Exempt

When a user has the Barge-in Exempt service enabled, another user (using the DPUBI service) cannot barge in on their calls. If a user attempts to use DPUBI to barge-in, then the barge-in is rejected and the user gets reorder tone.

If the Barge-in Exempt service is disabled, then DPUBI barge-in attempts are allowed as normal.

If a user has the Barge-in Exempt service enabled but has a single incoming, alerting call, then the call can be picked up by another user using the DPUBI service. Barge-in Exempt does not block pickup attempts.

Provisioning

The administrator can turn the warning tone on and off for each user. The users can see the status of the warning tone but cannot change it.

Users can activate and deactivate their Barge-in Exempt service via their web portal.

The administrator can select the Directed Call Pickup with Barge-in Feature Access Codes that apply to the group.

3.2.17 Distinctive Alert/Ringing

For details on Distinctive Alert/Ringing, see *Priority Alert*.

3.2.18 Diversion Inhibitor

The Diversion Inhibitor service prevents calls, redirected by a user, to be redirected again by the called party. This service is especially useful to help prevent calls from being answered by another user's voice mail when using Simultaneous Ring or Sequential Ring.

Description

The Diversion Inhibitor service allows a caller to inhibit redirecting services on the terminating side of an unanswered call, which is done using a feature access code (FAC). The FAC can be activated as a dial prefix on a per-call basis or as a static prefix for the destination number for the caller's redirection services.

Provisioning

The service is authorized at the service provider and group levels, and is assigned at the user level. The FAC is configurable via the web and OSS interfaces at the service provider and group level, and defaults to *80. There is no provisioning configuration for the service at the user level and it can only be used on a per-call basis via the FAC dial prefix.

This service is only available within a group or enterprise. For external calls outside of a group or enterprise, the prefix is ignored and the call is processed as usual (that is, redirection services are invoked).

The following services can be inhibited with the Diversion Inhibitor FAC:

Call Forwarding No-Answer, Busy, Always

Selective Call Forwarding

Voice Mail (Centrex Application Server Voice Mail and External Voice Mail)

Simultaneous Ring (Personal)

Sequential Ring

CommPilot Express

T

The following redirection services are not affected by the FAC and cannot be inhibited:

Remote Office

Hunt Group

Auto Attendant

Call Center

Call Pickup (all variations)

3.2.19 Do Not Disturb

This service allows a user to set their status as unavailable.

<u>Description</u>

When a user activates the Do Not Disturb (DND) service, all calls to the user are processed as if the user is busy and cannot receive calls. Other terminating services trigger on the busy condition, as if the user is really busy.

Since the usual busy processing applies to the call, the caller is unaware that the user has the service activated.

When active, the service provides a visual reminder to the user via a button on the CommPilot Call Manager. Furthermore, every time a call is blocked or deflected as a result of the service, the user is played a ring splash[4] as a reminder that the service is active.

<u>Provisioning</u>

CommPilot Personal Web Portal

The user can activate and deactivate the service through the CommPilot Personal web portal. The user can also select whether a ring splash is applied when a call is blocked or deflected by the service.

Feature Access Code

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[4] Users can activate and deactivate the ring splash reminder through the *Do Not Disturb* Provisioning page.

The user can activate and deactivate the Do Not Disturb (DND) service by dialling *78 (default) to activate or *77 (default) to deactivate. The system then plays a confirmation announcement and the user hangs up. Feature access codes can also be used from the CommPilot Call Manager. For example, the user can enter *78 to activate Do Not Disturb. The system then rings the user's device and plays a confirmation announcement indicating that Do Not Disturb has been activated.

CommPilot Call Manager

The user is provided with a shortcut to the Do Not Disturb CommPilot Personal web portal Provisioning page from the CommPilot Call Manager. In addition, a visual reminder of the status of Do Not Disturb is shown on the CommPilot Call Manager.

3.2.20 Malicious Call Trace

Malicious Call Trace is a user service that allows the service provider to trace calls coming to a Centrex Service Application Server user. When a user is assigned Malicious Call Trace, any call that attempts to terminate on that user triggers the generation of a report (or trace) that gets delivered to the service provider in an SNMP trap. The report contains information about the calling party (number, name), the time and date the call was received, and other relevant information (for example, redirection information).

Description

Malicious Call Trace (MCT) is a user service. Only a system administrator can manage (authorize, assign and, configure) this service. Note that service providers, administrators, group administrators, or end users cannot see the Malicious Call Trace feature in the CommPilot web portal (not even read only).

The system administrator can obtain a list of all Malicious Call Trace assignments on the system using the web portal, under the Service folder at system level, using the "Malicious Call Trace" link.

At execution time, the instant at which the alarm is generated depends on the trace type, as follows:

- Answered: The alarm is generated when the call is answered by the user being traced.
- Alerting: The alarm is generated when a call attempts to terminate on the user being traced, before any terminating services of the user are allowed to proceed.

- All: The alarm is generated when a call attempts to terminate on the user being traced, or when the user being traced originates a call (for example, upon sending an outgoing invite), including originations due to a redirection (such as call forward).

The alarm contains a subset of the Call Detail Record information. It contains all the available information at the time it is generated.

Provisioning

The system administrator specifies the following information once Malicious Call Trace is assigned to a user:

- Activation: On/Off
- Whether to trace for a time period only or not (applicable only when "On")
- Time period: Specified with a start date, a start time, stop date, stop time
- Trace type: One of the following three options:
  - Answered (all incoming answered calls) An alarm is generated for any terminating call that gets answered by the user being traced
  - Alerting (all incoming calls) An alarm is generated for any terminating call on the user being traced (answered or not)
  - All (all incoming and outgoing calls) An alarm is generated for any originating or terminating call involving the user being traced

3.2.21 Priority Alert

This service enables a user to define criteria to have certain incoming calls trigger distinctive alerting.

Description

The Priority Alert service allows a user to have some incoming calls alert them distinctively when meeting pre-specified criteria. The service applies to power ringing and alerting tones. In both cases, incoming calls meeting the criteria result in a distinct ringing cadence and alerting tone pattern, respectively. The distinctive alerting pattern is the same for ringing and tones. Apart from the distinctive alerting pattern, this service does not change the way incoming calls are processed.

This list of criteria includes:

Selected time schedule, for example, “Every Day All Day”

Whether the calling line ID is PRIVATE or UNAVAILABLE

A list of up to 12 phone numbers or digit patterns (for example, 514*)

Intra/inter-group status of the call

The criteria can be combined within predicates (for example, incoming call from this number AND within business hours AND during work week). Multiple predicates can be defined and distinctive alerting is provided when at least one of the predicates is met.

The service can also be assigned to Hunt Groups and Call Centers. In this case, the analysis of the incoming call against the set of criteria is done at the Hunt Group or Call Center level, and then affects the power-ringing pattern of all agents in the group. The Priority Alert feature does not need to be assigned to the agents themselves.

Provisioning

This service is configured through the CommPilot Personal web portal. A user can define criteria based on the incoming caller identity, ranges of digits, the time of day, the day of the week, and the inter/intra-group status of the call.

Ranges of digits can include digits from 0-9, and the following wildcard characters:

- ** (star)* – This wildcard can only be used as the last character of the digit string and matches any number of trailing digits.
- *? (question mark)* – This wildcard can be used anywhere in the string and matches any single digit.

Multiple criteria can be combined to build predicates, and multiple predicates can be defined simultaneously. Each predicate can be active or inactive. This service applies if at least one of the active predicates is met.

3.2.22 Selective Call Acceptance

This service enables a user to define criteria that allows incoming calls. All calls not meeting the specified criteria are rejected.

Description

Selective Call Acceptance allows a user to only accept calls that meet user-configurable criteria. Other calls are provided busy processing (for example, voice mail) or a system announcement..

The possible criteria include:

Selected time schedule, for example, “Every Day All Day”

Whether the calling line ID is PRIVATE or UNAVAILABLE

A list of up to 12 phone numbers or digit patterns (for example, 514*)

The criteria can be combined within predicates (for example, incoming call from this number AND within business hours AND during work week). Multiple predicates can be defined and any call meeting any predicate is allowed to terminate to the user. All other calls are rejected, and provided a busy condition or a system announcement.

Provisioning

The service is configured through the CommPilot Personal web portal. A user can define criteria based on the incoming caller identity, ranges of digits, a time schedule.

Ranges of digits can include digits from 0-9, and the following wildcard characters:

- ** (star)* – This wildcard can only be used as the last character of the digit string and matches any number of trailing digits.
- *? (question mark)* – This wildcard can be used anywhere in the string and matches any single digit.

Multiple criteria can be combined to build predicates, and multiple predicates can be defined simultaneously. Each predicate can be active or inactive. Incoming calls are accepted if at least one of the active predicates is met.

3.2.23 Selective Call Rejection

This service enables a user to define criteria that causes certain incoming calls to be rejected. All other calls terminate as usual.

Description

Selective Call Rejection allows a user to block calls that meet user-configurable criteria. The blocked calls are provided busy processing (for example, voice mail) or a system announcement. All calls not meeting the user-specified criteria are allowed to terminate normally.

The possible criteria include:

Selected time schedule, for example, “Every Day All Day”

Whether the calling line ID is PRIVATE or UNAVAILABLE

A list of up to 12 phone numbers or digit patterns (for example, 514*)

The criteria can be combined within predicates (for example, incoming call from this number AND within business hours AND during work week). Multiple predicates can be defined and Selective Call Rejection is provided if at least one of the predicates is met. All other calls terminate as usual.

Provisioning

The service is configured through the CommPilot Personal web portal. Through the portal, the user can define criteria based on the incoming caller identity, ranges of digits, and a time schedule.

Ranges of digits can include digits from 0-9, and the following wildcard characters:

- ** (star)* – This wildcard can only be used as the last character of the digits string and matches any number of trailing digits.
- *? (question mark)* – This wildcard can be used anywhere in the string and matches any single digit.

Multiple criteria can be combined to build predicates, and multiple predicates can be defined simultaneously. Each predicate can be active or inactive. Incoming calls are rejected if at least one of the active predicates is met.

3.2.24 Sequential Ring

This service allows users to define a “find-me” list of phone numbers or URLs, which are alerted sequentially upon receiving an incoming call that matches a set of criteria. While the service searches for the user, the calling party is provided with a greeting followed by periodic comfort announcements. The caller can also interrupt the search at any point to leave a message by pressing a DTMF key.

Description

When the Sequential Ring service is active on an incoming call, it takes control of the call and provides the calling party with an announcement stating the system will attempt to locate the user. This announcement is provided on a one-way voice path to the called party with a provisional response (183). No answer supervision is sent back to the caller (unless the incoming call was through an INVITE without SDP, in which case the Application Server has no choice but to answer the call in order to get an SDP from the caller).

The service then attempts to call the user by calling the phone numbers or URLs in the Sequential Ring list (starting with the user's base location, if enabled) one after the other until the call is answered, or the last number remains unanswered. At this point, the caller is provided with no-answer processing (CFNA, voice mail).

For each phone number in the Sequential Ring list, the following occurs:

- A call is originated to the phone number or URL and a timer is started. The timer is configured separately (in number of rings) for each location. Note that the number of rings for the base location is shared with the other "no-answer" services such as Voice Messaging and Call Forwarding No-Answer. Changing it in one service affects its value for the other services as well.

- If the called number is busy or results in a local announcement, the call is released and the service moves on to the next number. The base location can be configured so that Sequential Ring will not attempt any further location if the base location is busy. If this happens, busy processing will occur immediately. If all locations are, busy processing occurs as well.

- If the called party answers, the calling party is connected to the called party and the service ends.

- If the timer expires before the call is answered, the call is released and the service moves on to the next number.

- If this is option is enabled and the caller presses the # key, the search process is interrupted and the caller is presented with no-answer processing immediately.

Note that nothing prevents a destination from being configured (and thus alerted) twice, or the user from entering his/her own number should he/she wants his/her base location to be alerted last, for example

The Sequential Ring user defines a set of criteria that determine if the Sequential Ring service should be activated for the incoming call.

The set of criteria is analogous to the Selective Call Forwarding set of criteria and allows for defining a time schedule (time-of-day, day-of-week) and calling number(s) for which the service should be activated. If the criteria are met, the service is activated as described above. Otherwise, the call is processed as usual.

For Sequential Ring to always be enabled, for example, one criterion set to “Every Day All Day” and “Any phone number” must be defined and activated.

While the service goes through the list of locations, the caller is provided with the initial greeting followed by local ringback. Since the time necessary to find the user can be considerable (up to six times six rings, or more than three minutes), a comfort announcement is played after every 20 seconds of ringback.

Once a call is successfully connected, or when the last location in the list remains unanswered, the ringback or announcement is interrupted and the caller is connected to the user, or provided with no-answer processing, as applicable.

The feature is assigned like any ordinary user feature. It can be configured via the OSS interface by the system administrator or via the web interface by the user (or any administrator).

<u>Provisioning</u>

The following elements can be configured for Sequential Ring:

- o Whether to use the base location or not;
- o The number of rings for the base location (shared with other services such as CFNA);
- o Whether to continue searching if the base location is busy;
- o Whether the caller can press # to interrupt the search process or not;
- o The list of up to five locations and their timers (number of rings);
- o A list of criteria (similar to t the other “selective” services) and whether each one is active or not.

It is possible for the user to change the number of rings required for No-Answer handling by dialing a feature access code (FAC) code from his/her phone.

1) From a plain phone:
- o User dials *610.

- User listens to the prompt “Please enter the number of rings before the system applies No-Answer handling for your incoming calls”.
- User punches one digit (valid digits are 0, 2, 3, 4, 5, 6).
- User hears the announcement “Thank you. The system will apply No-Answer handling after… ‘n’ …rings for your incoming calls”.

2) From a SIP phone or call manager, the user can still use the FAC as stated above, or also:

- User dials *610*n#.
- User hears the announcement “Thank you. The system will apply No-Answer handling after… ‘n’ …rings for your incoming calls”.

When the value of *number of rings required for No-Answer handling* is set to “0”, the system will generate a No-Answer condition before the Voice Mail service sees the incoming call. As a result, the No-Answer handling will apply.

When the sequential ring number is configured, and validation option is enabled on system level, it is validated against the users calling plan profile and may be rejected.

3.2.25 Simultaneous Ring Personal

This service enables a user to have multiple destinations ring simultaneously when any calls are received on their phone number. The first destination to be answered is connected.

Description

The Simultaneous Ring Personal service is a user “find-me” service that alerts multiple terminating locations simultaneously. A user can provision up to ten secondary terminating phone numbers (for example, cell phone, home phone).

When a party calls a Centrex user, service issues simultaneous termination requests to the locations specified. The first location to answer the call is connected to the originating party; all other terminations are released.

All calls to secondary locations are subject to the services that apply to these locations. For instance, a call to a busy cell phone may get forwarded to voice mail, thus resulting in the other legs being released.

If all call legs are busy, the caller gets busy processing.

To avoid overwhelming a user when they are using their main location, the service can be configured to not alert the secondary phone numbers when the primary phone number is active on a call.

The secondary phone numbers can be any valid phone number or URL that is allowed by the Outgoing Calling Plan of the user.

Provisioning

The service is configured through the CommPilot Personal web portal. The user can:

- Activate or deactivate the service
- Enter up to ten secondary phone numbers
- Select whether secondary phone numbers or URL should be alerted while the primary location is active on a call

When the simultaneous ring number is configured, and validation option is enabled on system level, it is validated against the users calling plan profile and may be rejected.

3.3 End-User Services, Call Leg independent

3.3.1 Busy Lamp Field

The Busy Lamp Field (BLF) service enhances Centrex Application Server to support a SIP phone-based attendant console. It allows monitoring the hook status and remote party information of users via the busy lamp fields and appears on an attendant console phone.

It enables SIP attendant console phones to subscribe to a list of resources (users) to monitor, and receive notifications of the state of the monitored resources.

Description

The BLF service is a user service that supports the provisioning of an ordered list of monitored users, and a SIP URI on this list.

The list SIP URI addressing must be on a domain available to the user, and be unique amongst other list URIs within the system.

The order of monitored users corresponds to the line appearance order of the monitored user on the SIP attendant console phone. The initial NOTIFY for the subscription contains the full state of all the resources in the order specified in the BLF service configuration. If the monitored user list is modified (by adding, removing, or moving members) after there is an active subscription to the list, the subscription is terminated and Centrex Application Server sends a termination NOTIFY to the Attendant console phone. The phone should re-subscribe if needed.

Provisioning

The list of available users to be monitored is determined by the users within the enterprise or group of the user who is assigned the BLF service. If the user is not a member of an enterprise, then the available user list is taken from the group. The maximum number of monitored users that can be provisioned is fifty.

3.3.2 Calling Plan

The Calling Plan service allows the administrator to restrict the type of calls users can make and receive.

Description

The Calling Plan service allows the administrator to control the type of calls made and received by users in a group. The restrictions are applied by means of sets of call screening digit strings or complete E.164 and/or MSISDN numbers assigned to groups, departments, or single users

The administrator can define different screening by use of digit strings or complete E.164 and/or MSISDN numbers for outgoing and incoming calls. The following sub-sections describe these capabilities in more detail.

Call Topology

The administrator can define different screening numbers for the following:

- Outgoing calls – The outgoing call screening numbers allows the administrator to define how calls originated by users should be restricted.
- Incoming calls – The incoming call screening numbers allows the administrator to define how calls received by users should be restricted.
- Forwarding/transferring calls – The forwarding/transferring call screening template allows the administrator to define how calls that are redirected by the user services should be restricted.
- Being forwarded/transferred This call screening template allows for preventing users from being forwarded or transferred to external parties so to offer Originating Fully Restricted functionality. When an outgoing call is denied based on this call screening template, the user receives the applicable OCP treatment.

The call screening numbers apply independently to different legs of the call.

Call Types

The Incoming Call Screening numbers can screen the following call types:

- Calls from within the group – When this option is checked, users are allowed to receive calls from other members of the group.

- Calls from outside the group – The “calls from outside group” screening criterion of the ICP service provides a distinction between:

  A. Allow calls from outside of the group.

  B. Block calls from outside of the group

For a user, setting the “call from outside group” option to N disallows incoming calls from callers outside of the group, independently of how the call got to the user.

When an incoming call is denied by this new attribute, the caller receives the standard ICP denial announcement.

The Outgoing Call Screening numbers can screen the following call types:

Group – Calls from within the user's business group.

Local – Calls within the same geographic region.

International – Chargeable calls to other countries.

The same call types can be screened by the Forward/Transfer and the Being Forwarded/Transferred Call Screening templates.

The system administrator defines a digit map for the system that defines digit strings that should be mapped to each call type.

In addition to fixed call types, the Calling Plan service allows the administrator to screen calls against configurable digit strings. The digit strings are entered as fixed digit strings (for example, 2022517151) or digit patterns (for example, 202251*).

The administrator can define as many digit strings as required, and selectively assign them to the group, to selected departments or selected users. The digit strings can be used to complement the Outgoing Call Screening E.164 and/or MSISDN numbers.

Basic and Enhanced Screening Options

The Calling Plan service offers basic and enhanced screening options. The enhanced screening options apply only to the Outgoing Call Screening E.164 , MSISDN numbers and digit strings. The Incoming Call Screening digit strings, E.164 and/or MSISDN numbers remain the same with either option.

With the basic screening option, any outgoing call that is intercepted by the Calling Plan service is sent to an announcement informing the caller that the call is not allowed. Otherwise, the call is allowed to go through as usual.

With the enhanced screening options, the administrator can select how to process the calls that are intercepted by the service. The following interception options are offered for each call type or digit string:

- Allow – The call is allowed to proceed as usual (same as basic).
- Block – The call is routed to an announcement (same as basic).
- Authorization code – The caller is prompted for an authorization code. If a valid code is entered (through DTMF digits), the call is allowed to go through[5]; otherwise the call is blocked as described above.
- Transfer 1/2/3 – The caller is transferred to a configurable destination for further processing (for example, an attendant position). Three possible transfer destinations can be defined.

The enhanced screening options are only available when the Enhanced Outgoing Calling Plan (OCP) service is authorized and assigned to the group.

Sustained Authorization Codes

The Sustained Authorization Codes (SAC) feature allows users to unlock their calls by having their Calling Plan service use a sustained authorization code instead of prompting for the code on a per call basis. Users can also disable the sustained authorization codes feature, which restores the collection of authorization codes for each call.

Once a user has unlocked his/her calls, any call originated by a phone belonging to that user and for which the Calling Plan service would usually require the user to enter an authorization code is allowed to complete directly, without prompting the user for an authorization code. Instead, the code entered as part of the unlocking procedure is used implicitly, and captured in the call detail record (CDR) associated with that call.

Provisioning

The group administrator configures the Calling Plan service in a hierarchical fashion.

[5] The authorization code entered by the user is also captured in the accounting call detail record generated by Centrex Service Application Server.

- The administrator first defines default calling plans for the entire group. This plan applies to any department or user who does not have its own plan defined.

- The group administrator can define specific calling plans for selected departments in the group. The department calling plans have precedence over the group default calling plans for all users belonging to that department.

- The group administrator can define a specific calling plan for selected users. In this case, the user's calling plans have precedence over the department and group calling plans.

The Calling Plan service is configured through the CommPilot Group web portal. The configuration data is the same at each of the group, department, and user levels.

The following information items must be configured at each level when using the basic screening option:

- Define digit strings that should be screened if required. Digit strings can include digits from 0 through 9, and the following wildcard characters:

  - – * *(star)* – This wild card can only be used as the last character of the digits string and matches any number of trailing digits.

  - – *? (question mark)* – This wild card can be used anywhere in the string and matches any single digit.

- Allow/block for each system call type.

- Allow/block for each digit string.

This configuration must be done for outgoing calls. The incoming plan can also be configured to block or allow external calls. This process should be repeated for each department and user needing a specific calling plan.

When using the enhanced screening option, the configuration is similar, but the following information must also be configured as part of the outgoing calling plan to support the enhanced interception options:

Allow block/authorization code/transfer1-3 for each system call type.

Allow/block/authorization code/transfer1-3 for each digit string.

Provision the valid authorization code for the group, department, or user (each group, department and user profile uses its own list of valid authorization codes).

Provision the required transfer destinations (each group, department, and user profile uses its own list of redirection destinations).

NOTE: When creating a new department or user calling plan template, the default values for all configurable items are inherited from the layer above, which can be refined as required.

3.3.3 CommPilot Call Manager

The CommPilot Call Manager enables a user to use a web-based tool for service invocation and call control.

Description

The CommPilot Call Manager provides an alternative to pressing the flash hook and using star codes. It provides the user with a visual, graphical user interface to initiate, manipulate, and release calls.

The CommPilot Call Manager provides the following functions:

- *Navigation/support/help* – Useful links to help the user with the CommPilot Call Manager including support (to send an e-mail to the applicable support service), help (to display a context-sensitive help web pages), and configure (to jump to the CommPilot Personal web portal).
- *User information* – Presents the name, phone number, and extension of the user of the CommPilot Call Manager.
- *Service link area* – Provides status and Provisioning for commonly used services.
- *Call control* - Provides the user with call dispositions to make, control, and release calls by clicking them with the mouse (see below).
- *Dial* – Allows the user to initiate an outgoing call.
- *Call display* – Presents the user with information on active calls and allows the user to select calls with the mouse.

- ○ *Directories* – Provides access to the user directories (see below).
- ○ *Preferences* – Allows the user to configure the CommPilot Call Manager (see below).
- ○ *Outlook integration* – User has access to Outlook contacts, journaling, and vCards.
- ○ *Web pop-up* – User can configure a web server to be queried with the CLID of the caller to return a web page with related information.

The CommPilot Call Manager can be used in conjunction with any Centrex Service Application Server controlled device while maintaining a consistent behaviour regardless of device type.

Call Control

The following functions are provided for call control:

- *Click-to-dial* – enables a user to dial an entered number, dial from a phone list or Outlook, or redial the last called number.
- *Answer* – enables a user to answer a waiting call while already engaged in a call or to retrieve a held call.
- *Hold/retrieve* – enables a user to place an existing call on hold and then retrieve the call to resume conversation.
- *Release* – enables a user to disconnect a call that has been answered.
- *Transfer* – enables a user to redirect a ringing, active, or held call to another destination or directly to voice mail.
- *Conference* – enables a user to connect two existing calls into a three-way conference.
- *Forced off-hook support* – enables a user to Dial and Answer with a single click on the Call Manager by connecting to the phone in hands-free mode[6].

Directories

The CommPilot Call Manager provides the user with directories that can be used to make outgoing calls and obtain information on parties. These directories include:

- *Group* – The user can browse a list of other members of the group and call them directly from the list. Furthermore, the user can print the group directory in summary or detailed format directly from the CommPilot Call Manager.
- *Personal* – The user can create and maintain a list of personal contacts and call them directly from the list.

[6] This capability is available to users with CPE that support Answer-After and Talk SIP event package.

- *Call logs* – The user has access to a log of the last calls dialled, received and missed and can call them directly from the list.

- *Outlook contacts* – The user has access to their Microsoft Outlook contacts and can call them directly from the list. Furthermore, the user can configure the CommPilot Call Manager to have a matching vCard be presented automatically upon receiving a call.

LDAP – The users have access to an external LDAP contact database and can originate calls directly from the LDAP directory listings.

Service Status and Hyperlinks

The CommPilot Call Manager provides the user with direct access to the status of some commonly used services and presents their activation status on the CommPilot Call Manager. These services include:

- *Call Forwarding Always* –User can see the current status and have direct access to the Provisioning page.

- *Do Not Disturb* - User can see the current status and have direct access to the Provisioning page.

- *Remote Office* - User can see the current status and have direct access to the Provisioning page.

- *CommPilot Express* – User can see the current status and configure the state directly from the CommPilot Call Manager.

Outlook Integration

The CommPilot Call Manager makes use of Microsoft Outlook to provide the user with:

- *Contacts* - Access to the user's contact database for directories and dialling.

- *Journaling* – The user can have incoming and/or outgoing calls logged into the Outlook journal with their start time, answer time, and stop time.

- *vCard* – Users can bring up the vCard of other parties involved in calls (when available) and create new vCards for parties who do not have one already.

The integration with Outlook is done automatically and is transparent to the user. It does not require any Provisioning changes to Outlook.

The Outlook Integration service is supported for Outlook 2000 and later versions.

Web Pop-Up

The user can configure the CommPilot Call Manager to issue HTTP queries to an external server for active calls.

By clicking an icon, the user triggers an HTTP query to a specified URL. The URL is created from a string previously configured by the user. The URL string may contain static and dynamic fields. The dynamic fields are instantiated from the characteristics of the user and the call (for example, name, number, group, and so on).

Provisioning

The user can configure the CommPilot Call Manager as follows:

- The user can allow or disallow one-click dialling from the directories.
- The user can configure a web-pop upon incoming calls.
- The user can open an Outlook Journal entry for both incoming calls and/or outgoing calls.
- The user can select the Outlook Contact folder to use.

3.3.4 CommPilot Express

CommPilot Express enables a user to pre-configure multiple profiles for managing incoming calls differently based on a preset status as follows:

- Available – In the office
- Available – Out of the office
- Busy
- Unavailable

Description

CommPilot Express is a meta-service that consolidates Centrex call termination services into four profile-based call management templates. Each profile includes preferences for managing the relevant incoming call functions (for example, Call Forwarding [busy, no answer, always, selective], Voice Messaging, Simultaneous Ringing), which can be configured through a single easy-to-use web page.

The following profiles are defined:

- *Available - In the office* – This profile is meant for users working from their desk where their Centrex device is located. In this context, users need calls to be delivered to their regular device and optionally to another number or URL in case they are temporarily away from their desks (for example, mobile phone). Furthermore, users also need their incoming calls to be redirected to voice mail when they are busy, or unable to answer the call. Alternatively, users can choose to have their calls redirected to a selected phone number or URL instead (for example, an auto attendant or administrative assistant).

- *Available - Out of office* – This profile is meant for users working away from their desk for an extended period of time. In this case, users are interested in getting all of their calls sent to their temporary location. Optionally, these users may want to keep track of all incoming calls so they know if they missed some calls or simply want to keep a log of these calls so they can follow up on them when back in the office. For that purpose, this profile allows users to optionally specify an email address where a notification of all calls should be sent.

- *Busy* – This profile is meant for users who are temporarily unavailable to take calls, for instance when they are in a meeting. In this context, users are interested in screening their calls so only the most important ones come through. Hence, the busy profile allows the user to select up to three parties for which calls are allowed to come through. Other calls are sent directly to their voice mail. Furthermore, users may be interested in being notified of messages being left in their mailbox. So the “Busy” profile allows users to optionally specify an e-mail address where e-mail notifications should be sent.

- *Unavailable* – This profile is meant to be used outside of business hours, or while users are on vacation or holidays. In this case, users are interested in sending all their calls directly to voice mail or to a specified phone number or URL (auto attendant or administrative assistant), and provide callers with a distinctive greeting informing them of their unavailability, or regular business hours. Also, these users may be interested in having some critical calls come through anyway. Hence, the “Unavailable” profile allows users to send all calls directly to voice mail, and optionally allows them to specify a distinctive greeting and a list of up to three parties who should be allowed to alert the users at a specified phone number or URL.

Provisioning

CommPilot Personal Web Portal

The CommPilot Express templates are configured through the CommPilot Personal web portal. A user can also use the portal to select their active profile.

CommPilot Call Manager

A user can select their active CommPilot Express profile through a pull-down menu on the CommPilot Call Manager.

CommPilot Voice Portal

Users on the road or without web access can call into their voice portal to select their active CommPilot Express profile.

3.3.5 CommPilot Personal Web Portal

This service allows users to configure and customize services.

Description

The CommPilot Personal web portal provides users with a web interface that allows them to view, configure, activate, and deactivate their services.

The CommPilot Personal web portal also provides the users with:

- List of user services they are subscribed to.

- List of group services they are subscribed to.
- Context-sensitive help for every service.
- Feature access codes that are associated with subscribed to services.

The CommPilot Personal web portal is authenticated for each user with a user ID and a password, and provides a secure connection to the Centrex Service Application Server.

Login Wizard

When a new user or administrator attempts to log in for the first time, The Centrex Application Server detects the initial login attempt and redirects the user to a *web* page where the user can change his or her password.

When a user attempts to log in with an expired password, The Centrex Application Server also redirects the user to the same page to change the password. Previously, users with expired passwords were completely blocked from proceeding.

In both cases, the Centrex Application Server detects a valid user name and password. The user's identity is authenticated so the Centrex Application Server can provide the proper visual branding for the *Password Change web* page. However, the user is not authorized to do anything except change his or her password. The user is unable to navigate to any other pages within the web portal. Attempting to navigate to other pages by typing URLs directly into the browser address bar will fail. In addition, the Call Manager and Attendant Console windows do not automatically pop up until after the password has been changed.

The *Password Change web* page contains a form where the user enters his or her old password and new password. Submitting the form does two things:

1) It changes the user's password.

2) It redirects the user to whatever page would normally be displayed after a successful login.

<u>Provisioning</u>

The user can access the CommPilot Personal web portal from any standard web browser to perform service Provisioning.

3.3.6 Direct Inward/Outward Dialling

Users are assigned a public phone number that can be used to place or receive calls directly, without forcing access via a central number.

Description

Any Centrex user can be assigned a public phone number (also known as a Direct Inward Dialling phone number, or DID) that can be used by external parties to call the user directly, without having to go through an attendant. The Centrex Service Application Server provides the required translation, routing, and location policies to terminate inbound calls to the user associated with the dialled number.

Similarly, Centrex users can make outward calls directly without having to select an outbound access trunk or going through an attendant. The Centrex Service Application Server provides the required translation, routing, and location policies to originate outbound calls to any public phone number.

Provisioning

There are no Provisioning parameters.

3.3.7 Feature Access Code Service Chaining

Description

Feature Access Code Service Chaining enhances the validation performed on phone numbers entered on *configuration* pages for various Centrex Application Server services. It allows users to enter feature access codes (FAC) and speed codes in addition to phone numbers and extensions.

For instance, it can be used to configure an Auto Attendant to go directly to a user's voice mail by prefixing the destination number by the "Direct Voice Mail Transfer" feature access code.

Provisioning

Feature Access Code Service Chaining is used to enter feature access codes (FACs) and speed codes for the destination address for specific Centrex Application Server services.

The following services have been modified to allow the use of FACs and/or speed codes:

Blind Transfer

Call Forwarding (Always, Busy, No-Answer, Selective)

Auto Attendant

Hunt Group

Call Center

Simultaneous Ring and Simultaneous Ring Family

Sequential Ring

CommPilot Express

3.3.8 Intercom

The Intercom service allows a user to call another subscriber, where the system requests that the destination subscriber automatically answer. This provides for intercom-like functionality. A user or administrator can specify an accept list and a reject list. These are used to screen incoming Intercom sessions.

The accept list indicates which users are allowed to call a station.

The reject list indicates which users are not allowed to call a station.

In both lists, a wildcard can be used, which indicates all stations.

Description

Intercom Origination

A user originates an Intercom call by dialing the Intercom FAC. If the users do not supply a destination address when dialing the Intercom FAC, then they are given stutter dial tone so that they can enter the destination address for the Intercom call.

- If no destination address is entered (that is, digit collection times out), then the call is sent to reorder treatment.
- If a destination address is provided, then an Intercom call is originated to the destination address.

NOTE: The Intercom origination is processed by the user's origination services such as the Outgoing Calling Plan.

When the originator of an Intercom call receives an indication that the call has been answered, an Intercom confirmation tone is played to both the originator and terminator. Once the confirmation tone has finished playing, the media path between the originator and terminator is established according to the Outgoing Connection Type.

If the user has the Outgoing Connection Type option set to One-Way, then no media can be transmitted from the terminator to the originator, after the call is answered. Only the originator is allowed to transmit media. If the terminator answers the call in order to play treatment, the originator will not hear the treatment since the call has been answered and the connection is now one-way.

NOTE: Before answer, the connection is always two-way so that remote media (such as remote ringback and early treatment) can be heard prior to answer.

If the Outgoing Connection Type option is set to Two-Way, then the originator and terminator can talk to each other as usual.

Intercom Termination

When users receive an Intercom call, it is screened using their Access List according to the Access List type setting.

If the Access List type option is set to "Allow calls from only the users selected below" (that is, an Accept List), then the Intercom call is rejected with the Intercom Rejected announcement, if the originator is not in the Access List.

If the Access List type option is set to "Allow calls from everyone except the users selected below" (that is, a Reject List), then the Intercom call is rejected with the Intercom Rejected announcement, if the originator is in the Access List.

Note that the Access List in this release can only contain other users in the group and/or enterprise. If the terminator has the Access List set to be an Accept List, then all Intercom calls from outside the group and enterprise are rejected since the originator cannot be in the list. Similarly, if the terminator has the Access List set to be a Reject List, then all Intercom calls from outside the group and enterprise are accepted since the originator cannot be in the list.

If the user's Access List allows the Intercom call, then it is allowed to continue. If the Auto-Answer option setting is set to "On", then a header like the following is added to the INVITE(s) sent to the user's SIP device:

```
Call-Info:<sip:as.broadsoft.com>;answer-after=0
```

This header is defined in the Centrex Application Server Advanced Call Control Specification and indicates that the device should immediately answer the call. If the device supports the header, then the Intercom call is automatically answered. If the device doesn't support the header, then the Intercom call must be manually answered.

Note that if the terminating user does not have the Intercom service assigned, then an incoming Intercom call is treated as a normal call termination instead of an Intercom termination (for example, no Auto-Answer, no Access List screening), but continues to be considered an Intercom call for service interactions.

Provisioning

The Intercom service is a user service. Therefore, it can be authorized to service providers/enterprises and groups, and can be assigned to users.

There are several configurable options for the Intercom service. All are configurable by both administrators and users.

The Outgoing Connection Type option can be set to One-Way or Two-Way. When set to One-Way, only the originator of a Intercom call can talk. When set to Two-Way, both the originator and terminator of an Intercom can talk. Its default setting is Two-Way.

The Auto-Answer option can be set to On or Off. When users receive an Intercom call with Auto-Answer set to On, their device is signaled to automatically answer the call. Its default setting is On.

The Access List type option can be set to “Allow calls from only the users selected below” (that is, an Accept List) or “Allow calls from everyone except the users selected below” (that is, a Reject List). Its default setting is “Allow calls from only the users selected below”.

The Access List itself is a list of users within the Intercom user's Group and/or Enterprise (if the user is a member of an Enterprise). Users can be searched for, added to, and removed from the Access List. The Access List is used as defined by the Access List type option.

At the group and service provider/enterprise levels, the Intercom FAC can be configured by administrators. It defaults to *50, but can be changed to any valid, available FAC.

3.3.9 Music On Hold User

This feature enables users to enable/disable Music On Hold on either a per-call or persistent basis by using a feature access code or the web portal.

Description

Music On Hold user configuration allows a user to turn on or off music for held and parked calls by the user when the group has enabled the Music On Hold for held and parked calls. This configuration is available to all users within a group who have Music On Hold assigned. This configuration is not available to virtual users and it is not applicable to users within a call center.

The user can configure this via the web portal or the OSS and also turn it off per call with a feature access code. Using the web portal, the user can configure this via a *Music On Hold* configuration user page, which is available as a link on the *User Call Control* page (or via the OSS using the modifyUserService command). If Music On Hold is turned off, then callers will hear silence for calls that are held and parked. If Music On Hold is turned on, then callers will hear for calls that are held and parked, (assuming Music On Hold is enabled for held and parked calls for the group). The user configuration is on by default when the user is created, or when the service is assigned to the group.

A new feature access code (FAC) called Music On Hold Per-Call Deactivation is added to the service provider and group. This FAC can be configured for a service provider when Music On Hold is authorized to the service provider using the existing service provider *Feature Access Code* page or the service provider modifyFacCode OSS command. This FAC can be configured for a group when Music On Hold is authorized to the group at the existing group *Feature Access Code* page or the group modifyFacCode OSS command. The default value for the FAC is *60.

A user can turn off music for all held or parked calls prior to dialing a call or when a call is in progress. It can be turned off prior to a call by dialing the FAC before dialing the outgoing telephone number. When the user dials the FAC, the user is provided with a confirmation tone followed by dial tone after which the user dials the outgoing telephone number. It can be turned off when a call is in progress by flashing the switch hook and then dialing the FAC. After the FAC is dialed, the system provides a confirmation tone followed by the dial tone. The user can now flash back to the active call. Music will be turned off for all user sessions until the user who initiated the FAC is disconnected. Once the call is disconnected the Music On Hold configuration will be returned to the persistent state.

Provisioning

The user activates and deactivates Music On Hold via the CommPilot personal portal (persistently) or via a feature access code (per call).

3.3.10 Remote Office

This service enables users to access and use their Centrex profile and services from any device, on-net, or off-net (for example, home office or mobile phone).

Description

Remote Office is especially useful for telecommuters and mobile workers, as it enables them to use all of their features while working remotely (for example, extension dialling, transfers, conference calls, Outlook Integration, directories, and so on). In addition, since calls are still originated from the Centrex Service Application Server, the service provides an easy mechanism for separating personal and business phone expenses, as well as keeping alternate phone numbers private.

To use the service, users simply enter the phone number of their current location and activate the service. From that point on, their usual Centrex location is temporarily overridden by the newly configured location.

When the service is active, all incoming call to users are redirected to their Remote Office location and are subjected to the user’s terminating services.

Similarly, a user can originate calls from their Remote Office location through the CommPilot Call Manager click to dial capability. This ensures that calls are processed by The Centrex Service Application Server as normal originating calls, and are subjected to the users’ originating services.

This service allows a user to manage active calls as usual through the CommPilot Call Manager, thus providing the user with their Centrex profile and services from any addressable phone on the network or the PSTN.

When the service is active, a reminder is provided on the user’s CommPilot Call Manager, indicating that incoming calls are redirected to an alternate destination.

For initiating calls using this service, it is required that the user is registered as the call will fail if the status of user is not-registered in HSS.

Provisioning

CommPilot Personal

A user can configure their temporary location and activate the service through the CommPilot Personal web portal. The location is entered as a phone number.

When the Remote Office number is configured, and validation option is enabled on system level, it is validated against the users calling plan profile and may be rejected.

CommPilot Call Manager

The user can click the Remote Office status indicator on the CommPilot Call Manager to link directly to the CommPilot Personal web portal Remote Office Provisioning page.

3.3.11 Web Portal Call Logs

The Web Call Logs service allows for storing several days of logs for each user, or a much larger number of logs than the 20 per-call type of Basic Call Logs. The maximum number of logs per user can be set independently for each enterprise or service provider.

<u>Description</u>

When the Web Call Logs feature is assigned, logs are to an external call detail server, using the Radius protocol. They are retrieved using the SOAP over HTTP protocol.

A new *web* page showing the users' call logs is added. This *web* page fetches all the call log information from the call detail server and displays it under three tabs, one for each call type (placed, received, missed). The number of call logs displayed depends on the enterprise/service provider's configuration.

<u>Provisioning</u>

For each enterprise/service provider, a maximum number of call logs per user (per-call type), along with an expiry period in days, are configured via the web portal.

The system administrator is responsible for configuring the enterprise/service provider values (the service provider admin only sees a read-only page).

3.4 Group Services, Originating Call Leg

3.4.1 Account Codes

The Account Code service allows the users to assign certain calls to specified accounts for tracking purposes.

<u>Description</u>

Two account code dialling methods are offered, which can be assigned concurrently to different users of a group.

Mandatory Account Codes

Users assigned the mandatory Account Codes service are prompted to enter an account code every time they make a call outside of the group. When prompted to enter a code, the user dials the applicable digit string, after which the call resumes normally. The code is captured in the associated accounting information generated for that call.

The group administrator can select to have account codes apply only to long distance calls. In this case, users are not prompted for an account code when making a local or toll-free call.

Account codes are a fixed length, as configured by the group administrator. When prompted for an account code, the user is informed of the number of digits to enter. Hence, when a user makes a call for which an account code is required, the dialling sequence is as follows:

[User dials phone number] [account code prompt] [user enters account code] call proceeds.

Emergency and repair calls are never prompted for an account code.

Feature Access Code (FAC)-based Account Codes

When assigned the FAC-based Account Code service, the user can:

- Make a call as usual, without entering an account code.
- Dial a feature access code (for example, *XX) before making a call. In this case, the user is prompted for an account code, dials the code, receives confirmation, and the proceeds with the call as usual. The sequence is as follows: [Off-hook] [FAC] [prompt] [code] [confirmation] [dial tone] [call]

- Flash the switch-hook during the call and enter a feature access code (for example, *XX). In this case, the user is prompted for an account code, dials the account code, and then is reconnected to the call. The sequence is as follows: [Call] [flash] [FAC] [prompt] [code] [confirmation] [call]

Note that the last two methods can be used concurrently, in which case the last account code to be entered is the one that is captured in the associated accounting information generated for that call.

Provisioning

The group administrator configures the Account Code service through the CommPilot Group web portal.

When configuring the service, the group administrator:

- Activates or deactivates the service
- Selects the length of the account code
- Selects whether the service should apply only to long distance calls
- Selects which users of the group are assigned the service
- Selects the activation method to be used for each user

Users can view the *Account Code* page but cannot modify it.

3.4.2 Authorization Codes

The Authorization Codes service allows the group administrator to select specific users who must enter a valid authorization code when making a call to a party outside of the group.

Description

Users assigned the Authorization Code service are prompted to enter a valid authorization code when making a call outside of the group. Unlike account codes, authorization codes entered by a user must match one of the valid codes previously configured by the group administrator.

When a user dials a number for which an authorization code is required, The Centrex Service Application Server prompts the user for a valid code. The user then dials the authorization code followed by the (#) (or waits for the inter-digit timeout). If the code entered does not match any of the valid authorization codes configured for the group, the user is provided with another attempt at entering a valid code. If the second attempt is also unsuccessful, the user is sent to an error treatment. If a valid code is entered, the call is allowed to proceed as usual and the authorization code entered by the user is captured in the accounting information generated for that call.

The group administrator can select to have account codes apply only to long distance calls. In this case, users are not prompted for an account code when making a local or a toll-free call.

Emergency and repair calls are never prompted for an authorization code.

Provisioning

The group administrator configures the Authorization Code service through the CommPilot Group web portal.

Through the Provisioning page, the group administrator can:

- Activate or deactivate the service
- Select the length of the authorization codes (2 to 14)
- Select whether non-toll calls are subject to an authorization code
- Configure valid authorization codes (and optional descriptions)
- Select which users in the group are required to use the authorization code service

Note that the Authorization Code service applies to all applicable calls made from a device assigned to a user who is subject to the service.

3.4.3 Call Intercept

This service allows intercepting calls routed to a line that is not manned for some time with an informative announcement and alternate routing options (for example, “This number is for the time being not manned. To talk to an operator, press 1”).

The service may be used to decommission a single user or an entire group.

Description

If the Call Intercept service is assigned to a user or a group, then incoming calls to the user are intercepted and played an announcement. If configured, this announcement plays back a new destination number to the caller and offers the caller to connect to this new number.

The announcement can be in audio or video format, depending on the service configuration and the calling party's ability to support video.

Outgoing calls are prohibited from a user with the Call Intercept service assigned. Only emergency and repair calls are permitted. All other call attempts are rejected and the user is played a treatment.

Provisioning

The service provider uses the CommPilot Service Provider web portal to authorize and assign the Call Intercept service to a group.

The group administrator assigns the Call Intercept service to selected users using the CommPilot Group web portal.

The Call Intercept service allows three types of incoming call interception:

- Static – An out-of-service announcement (audio or video) is played twice followed by a fast-busy treatment.
- Hear new destination – The out-of-service announcement is complemented with the playback of the user's new phone number.
- Connect to new destination – After hearing the new phone number, the caller can press a digit to be immediately transferred.

The desired type of interception is configured when the service is assigned to a group or a user.

When transfer on "0" to new phone number is used, the redirection number is configured and validation option is enabled on system level, it is validated against the users calling plan profile and may be rejected.

3.4.4 Call Park

The Call Park service allows a user to suspend a call for an extended period of time. During this time, the user can freely make and receive other calls and invoke other features without limitation. When ready, the user can retrieve the parked call from any extension.

Description

The Call Park service allows users to park a call so that any member of the group can retrieve it with the Call Park Retrieve function.

A call can be parked against any user of the group, including the user who parks the call. However, a user can only have one call parked at a time.

To park a call, the user presses the flash-hook during an established call and then dials the Call Park feature access code, after which the user is prompted to enter a number[7] then the call is parked. If no number is entered and the user hangs up immediately after dialling the feature access code, the call is parked against the user's line.

Once a call is parked, it no longer appears on the user's Call Manager and the user can hang up or perform other telephone tasks.

While parked, the parked parties hear the audio on hold configured for that group[8].

To retrieve a parked call, users dial the Call Park Retrieve feature access code, which results in prompting the users to enter a number[9] where the call to be retrieved is parked. Upon entering the number, users get reconnected to the parked party. If no number is dialled after the feature access code, the user is reconnected to the call parked against his/her line[10].

A 45-second timer is started when a user parks a call. If the timer expires before the parked call is retrieved, the Centrex Service Application Server determines if the parking party is idle. If so, the parking party is alerted and the call appears on the parking party's CommPilot Call Manager as a held call, and the user's phone is rung (if on-hook). The behaviour is similar to hold recall.

If the parking party is not idle, the timer is restarted for 10 seconds and the call remains parked. This procedure is repeated until the parking party can be alerted or the parked call is retrieved or released.

Call Manager

[7] Although entering a full E.164 number is supported, the party against which the call is parked must be in the same group as the party parking the call.
[8] If no audio on hold is configured for the group of the user parking the call, the parked party hears silence.
[9] Although entering a full E.164 number is supported, the party against which a call is parked must be in the same group as the party parking the call.
[10] To retrieve calls parked against them, users enter the call park retrieve feature access code followed by the #, an inter-digit timeout, or their own extension.

Calls can be parked and retrieved through the CommPilot Call Manager. To park an active call, the user enters the Call Park feature access code in the Dial window of the Call Manager and then clicks the Dial button. This results in connecting the user to the Call Park number prompt. The user can then resume the Call Park interactions on the phone itself.

Similarly, a user retrieves calls by entering the Call Park Retrieve feature access code in the Call Manager Dial window and clicks Dial while idle, or while involved in another call. This results in connecting the user to the Call Park Retrieve prompt. The user can then resume the interactions on the phone itself.

Provisioning

The group administrator assigns the Call Park service to the entire group at once through the CommPilot Group web portal. Once assigned, all users in the group can park and retrieve calls.

The group administrator can also configure a default audio on hold for the group, which is played to all parked calls.

The group administrator also configures the Call Park service and Call Park retrieve feature access codes through the CommPilot Group web portal.

3.4.5 Calling Group ID Delivery

This service allows the group administrator to assign a Calling Line Identity (name and number) to an entire group.

Description

This service allows the group administrator to define a default group Calling Line ID (CLID). The default group CLID is made up of:

- *Name* – The name of the group (truncated to 15 characters).
- *Number* – A valid phone number authorized to the group, also known as direct-inward dialling (DID).

For users with their own DID, the administrator can select whether the default group name and/or number should override the users' own name and number.

In all cases, if the user making a call, blocked the delivery of the CLID, the presentation indicator sent to the far-end party is set accordingly and the presentation of the group CLID is blocked.

Provisioning

The following options are offered to the administrator to configure the Calling Group ID Delivery service:

- *Group number* – The administrator can select the group DID among the ones authorized for the group. Note that the selected DID is still available to be assigned to a real or virtual user of the group.

3.5 Group Services, Terminating Call Leg

3.5.1 Auto Attendant

The Centrex Auto Attendant provides enterprises with a powerful and flexible tool to field inbound calls and deliver them to the intended destination through interactions with the caller. The Centrex Auto Attendant is an integral part of the Centrex Service Application Server product offering and does not require an external third-party system.

Description

The Centrex Auto Attendant is reached by dialling an associated phone number or an extension. Once connected to the Auto Attendant, the caller is played a greeting that provides a menu of options to complete call routing.

The menu, which is configured by a group administrator, can provide up to nine options to the caller, including:

- *One-key dialling* – The caller presses a pre-defined DTMF key to reach a particular phone number or extension within the group. This option is also used to build multilevel IVR menus.

- *Operator dialling* – The caller presses a pre-defined DTMF key to reach an operator.

- *Name dialling* – The caller spells the name of the intended party using the numerical DTMF keypad. Upon identifying a unique match, the caller is played the name of the called party and is then transferred.

- *Extension dialling* – The caller enters the extension of the intended party through the numerical DTMF keypad. Upon collecting the full extension, the caller is played the name of the called party and is then transferred.

- *Immediate Extension Dialing* – The group administrator may elect to allow callers to dial an extension from the first-level menu. The First-Level Extension Dialing option allows the administrator to enable or disable immediate extension dialing for a given auto attendant. When the feature is enabled, the caller to the auto attendant can dial the desired extension right away on the first level of the auto attendant without having to first navigate to the second-level of the AA menu.

- *Dial by First Name* – The group administrator may elect to allow name dialing from a combined FirstName-LastName in addition to the current LastName-FirstName list.

- *Holiday Schedule* – A group administrator may define a holiday schedule that can be associated with an auto attendant. More than one holiday schedule maybe created. Each holiday schedule may be a maximum of 20 dates or date ranges.

- *Enhanced Business Hour Support* – A group administrator can define time schedules for their group. Multiple time schedules can be created. Time schedules consist of 20 date/time ranges for a week. Time schedules can be business hours, call center hours, after business hours, and so on. Time schedules created by the group are visible to groups and users.

Auto Provisioning Users in a Group

The moves, adds, and changes for users in a group are automatically available for the Auto Attendant name dialling and extension dialling functions. Access to users currently in the group is always available.

Multi-Site Support

The Auto Attendant supports geographically distributed user of a group.

Support for Users without DID

The Auto Attendant supports users with a direct inward dialing (DID) number as well as users without an external public directory number. These users originate calls as usual and the Auto Attendant allows them to receive external calls. Calls made to the Auto Attendant use the routing capabilities described above to terminate calls to the appropriate user.

This support provides greater flexibility for a group administrator to create and delete users and in many cases reduces the costs associated with obtaining DID numbers.

Video Support

The Auto Attendant can provide the caller with an audio or video menu, based on the Auto Attendant profile and the capabilities of the calling party’s endpoint.

<u>Provisioning</u>

The group administrator configures the Auto Attendant through the CommPilot Group web portal.

The following options are provided on the Attendant Provisioning screen:

- *Greeting* – The group administrator can select the default Auto Attendant greeting or upload a customized greeting that matches the available options.

- *Default menu options* – The group administrator can assign keys to the default menu options of the Auto Attendant (operator, name, and extension dialling).
  - The administrator can configure the auto attendant to allow extension dialling on the first level of menu.
  - The administrator can configure the auto attendant to allow name dialling with first name entered before last name,

- *Customized menu options* – The group administrator can create customized menu options by associating keys to specific phone numbers.

- *Customized actions* – The group administrator can assign specific actions to the key entered by the user.

- *Business Hours – The administrator can define the business hours for the auto attendant and select the business hours pattern that applies to the attendant via the web portal.*

- *Holiday schedule – The administrator can define holiday schedules via the web portal.*

- *Centrex Voice Portal greeting change* - The group administrator can record new greeting menus through the Voice Portal phone interface. This automatically provisions the newly recorded greeting as the active greeting for the Auto Attendant.

3.5.2 Call Centres

The Centrex Service Application Server provides support for basic call centres, allowing business agents to receive incoming calls from a central phone number. Using this service a business can establish technical assistance lines, customer support numbers, or order-taking centres. Multiple call centres can be supported per business. Incoming calls to a call centre are presented to the next available agent.

<u>Description</u>

The Call Centre service builds on the basic Hunt Group service to provide a complete, business-ready application. Hence, call centres inherit all of the characteristics of the Hunt Group service and are also provided with sophisticated call-handling features like queuing, music on hold, and so on. Refer to *Hunt Groups* for a complete description of Hunt Group service features and characteristics.

The following sub-sections describe the additional features provided by the Call Centre service.

Geographical Distribution

The Centrex Service Application Server expands the capabilities of legacy call centres by allowing call centre agents to be geographically distributed. Therefore agents can attend calls from home, a satellite office, or any other location served by the Centrex Service in a transparent fashion.

Features

The Centrex Service Application Server Call Centre functionality can be combined with other Centrex Call Completion services to ensure that all incoming calls are serviced expeditiously under any network condition and at anytime.

- *Voice mail* – If there are no agents to handle an incoming call or the call goes unanswered for a specified amount of time, the call can be forwarded to a call centre voice mailbox.

- *Night service* – Calls received after-hours or on non-business days receive a service menu of options allowing a caller to leave a voice message or transfer to an emergency number.

- *Multiple call distribution policies* – Incoming calls are handled according to the selected policy, which include uniform call distribution, linear hunt group, circular hunt group, simultaneous ringing and no-answer.

- *Call queuing* – When all call centre agents are busy, incoming calls can be queued until they can be presented to an available agent.

- *Queue escape* – Callers who are queued can press a key to be sent directly to the call centre voice mailbox instead of waiting for an available agent.

- *Overflow* – When a call centre cannot accept any more calls, incoming calls can be forwarded to an overflow phone number.

- *Statistics* – Statistics are generated for each call centre and each agent in the call centres, and can be viewed by the group administrator via the web portal and/or periodically dispatched to a configurable destination.

- *Service integration* – Any Centrex personal service can be assigned to a call centre phone number to customize the call centre group. This includes services such as Call Forwarding, Call Notification, Call Screening, and Voice Messaging.

- *Queue flushing* – When all agents in the call centre group log out, queued calls are automatically sent to the call centre group voice mailbox.

- *Outlook contact integration* – vCards from the agent's Outlook or Exchange contact database pop up for incoming calls.

- *Agent log in/log off* – Agents can log in and out from the group so that calls are only presented to agents who are on duty.

- *Screen pop-ups* – Incoming calls pop up on a web screen showing information associated with the incoming call. A group-specific URL is accessed for each call.

- Configurable Music on Hold – The queued callers are provided with an initial greeting, followed by music or advertisements and periodic comfort announcements. All announcements can be played in audio or video format, based on the call center profile and the capabilities of the caller's endpoint.

<u>Provisioning</u>

The group administrator configures call centres through the CommPilot Group web portal. Through the Provisioning pages, the administrator can configure:

- Users who should be part of the call centre (in order)

- Chose the Phone number of the call centre[11]

- Queue length

- Hunting policy (circular, regular, simultaneous, uniform)

- No-answer policy (applied in overlay of the other policy)

- Statistics dispatch, sampling period, and destination

- Included Phone number and extension

[11] The virtual users to use as Call Centre numbers are provided by the Service Provider to the IMS network and the Application Server.

- Queue greeting, comfort tone, and music on hold
- Delay between comfort tones
- Agent's ability to log in and out of the call centre

The administrator can also consult the call centre daily statistics that provide the following information:

- Current number of calls in queue
- Number of incoming calls to the group
- Number of calls queued for today
- Number of busy overflows for today
- Number of calls answered for today
- Average time spent with an agent
- Average time a call spent in the queue
- The number of calls processed by each agent

These statistics are provided for the current and previous day.

3.5.3 Call Pickup

Call Pickup allows users to answer any ringing line within their call-pickup group. A call pickup group is defined by the administrator and is a subset of the users in the group that can pick up each other's calls.

Description

To pick up a ringing call coming to another user of the group, users go off-hook and dial the Call Pickup feature access code, which connects them to the ringing party.

If more than one line in the call pickup group is ringing, the call that has been ringing the longest gets picked up.

Users already engaged in a two-way call can flash the switch hook to put the other party on hold and dial the call pickup feature access code to answer an incoming call to the call pickup group. Users then flash the switch hook to toggle between the two parties, or use the CommPilot Call Manager to control the two calls.

Call Manager

Users can pick up a call through the CommPilot Call Manager, either when idle or busy on one other call. They simply enter the Call Pickup feature access code in the Call Manager Dial window and then click the Dial button. This results in ringing the phone or placing the other party on hold, and connecting to the ringing party.

Provisioning

The group administrator defines Call Pickup groups through the CommPilot Group web portal. A single group can have multiple call pickup groups defined simultaneously, but a given user can only belong to a single Call Pickup group.

The group administrator also defines the Call Pickup feature access code through the CommPilot Group web portal.

3.5.4 Custom Ringback Group

The two Custom Ringback group services (one for audio and one for video) allows a user to specify custom media files to be used for ringback, when incoming calls are received. When the user is called, the system allocates a media resource and plays a custom ringback file to the caller instead of the standard ringback tone.

The user services allow a user to specify multiple profiles. Each profile is associated with a set of criteria (phone numbers, time of day, and so on) and a custom media file. When a call is received, it is compared with the profiles associated with the user. If a match is found, then the associated custom media file is used; otherwise the group service is checked. If active, then the group's custom media file is used, otherwise system ringback is provided.

Description

The Custom Ringback Group service allows an audio ringback at the group level to be defined.

When the Custom Ringback Group - Video service is assigned, it enhances the Custom Ringback Group service. It allows a video ringback to be configured in addition to an audio ringback at the group level.

NOTE: The Custom Ringback Group service can be turned on only when at least one custom ringback file (audio or video) is defined.

3.5.5 Hunt Groups

The Hunt Group service allows incoming calls to a central phone number to be distributed among the members of that group according to a hunting policy.

<u>Description</u>

The Hunt Group service allows processing of a high volume of calls to a single phone number by distributing the incoming calls to multiple users according to a hunting policy. Based on the chosen policy, an incoming call hunts for an idle user in the group to terminate the call to that user.

Hunting Policies

When a hunt group is created, the users are provisioned in a Hunt Group list. The hunting process essentially determines how to process that list to find an idle user where to terminate the call.

The Centrex Service Application Server supports the following hunting policies:

- *Regular (linear)* – The incoming calls to the group start hunting on the first user in the list and hunt all the provisioned users sequentially until an idle user is found or the end of the list is reached.

- *Circular* – The incoming calls to the group start hunting with the user following the last user to receive a call. When the end of the list is reached, the hunting circles back to the first user in the list. The hunting ends when an idle user is found or all the users have been visited.

- *Uniform* – The incoming calls to the group are presented with the user that has been idle for the longest time.

- *Simultaneous* – The incoming calls alert all idle users in the group. The call is connected to the first user to answer the call.

- Weighted – The incoming calls alert agents in a pseudo-random fashion according to their relative weight. Agents with a higher weight are assigned more incoming calls than agents with lower weights.

In all cases, if all users in the hunt group are busy, the incoming call is provided with the busy processing that applies to the hunt group.

Hunt Group Services

The Centrex hunt groups are assigned a phone number like regular users, and can also be assigned services like regular users. The following services can be assigned to hunt groups:

- *Call Forward Always* – This service redirects calls to the hunt group to the selected destination.
- *Call Forward Busy* – This service redirects calls to the selected destination if all the users in the group are busy.
- *Call Forward No Answer* – This service redirects calls to the selected destination if the user does not answer before the expiration of the timeout.
- *Voice Mail* – This service redirects calls to the group voice mail if all agents are busy or if the call remains unanswered for too long.
- *Selective Call Forwarding* – This service redirects calls to the hunt group based on the time-of-day, day-of-week, or the CLID of the caller.
- *Selective Call Acceptance/Rejection* – This service blocks calls to the hunt group based on the time-of-day, day-of-week, or the CLID of the caller.
- *Incoming Call Notification* – This service reports all incoming calls to the hunt group to the selected e-mail account.
- *Incoming Calling Plan* – This service allows selective blocking of external calls to the hunt group.
- *Anonymous Call Rejection* – This service allows blocking of anonymous calls to the hunt group.
- *Do Not Disturb* – This service makes the hunt group appear as busy.

User Services

Users who are member of a hunt group can have their own phone number where they receive calls, and have their own service independently of the hunt group services.

To maintain consistency of the hunting policy when traversing the list of users, the calls presented to the users by the hunt group are subject to the following service interactions:

- *Call Waiting* – When a member of the hunt group is busy but eligible for call waiting, it is considered available to receive a call by the hunt group.

- *Call Forwarding (all types)* – Incoming calls to the hunt group are never forwarded by any call forwarding service assigned to a member of the hunt group.

- *Voice Mail* – Incoming calls to the hunt group are never forwarded by the voice mail service assigned to a member of the hunt group.

- *Call Transfer* – When an incoming call is sent to an available member of the hunt group, this member cannot transfer or blind transfer this call to another party until it is answered.

Provisioning

An administrator configures the Hunt Group service using the CommPilot Group web portal[12]. There is no limit to the number of hunt groups that can be created in a group, and a given user can be part of more than one hunt group.

The following attributes can be configured when creating a hunt group:

- *Group name* – Allows for selecting the name of the hunt group. This name is prefixed to the caller ID delivered to the user when a call terminated through a hunt group. Hence, if a call from Bob Smith gets presented to a user through the "Support" hunt group, the CLID appears as Support – Bob Smith on the user's phone and Call Manager.

- *Group number* – Allows for selecting the number used to call the hunt group. This number must have been previously authorized for the group, and be available.

- *Extension* – Allows for selecting the extension that can be used by other members of the group to call the hunt group. By default, it is set to the last *N* digits of the group number, and can be modified by the administrator.

- *Group policy* – Allows for selecting the hunting policy used for that group, as defined in the previous section.

[12] The virtual users to use as Hunt Group numbers are provided by the Service Provider to the IMS network and the Application Server.

- *Members* – Allows for selecting the members of the hunt group among users of them group. The members are provisioned in an ordered list.

As described in the previous section, the administrator can also assign services to the hunt group through the CommPilot Group web portal.

3.5.6 Conferencing

This service allows end users to set up a multi-party audio conferences hosted on the Centrex Services Conferencing Server.

Description

Multi-port conference bridges can be used by members of the group and external parties to hold scheduled, recurring, reservation-less, and ad-hoc conferences.

Groups who subscribe to the Conferencing service are allocated a maximum number of conference ports that may be used simultaneously.

Groups will get conference numbers, the amount of numbers that shall be provided to group will be decided when the Conference number is created in the CAS system

Group administrators can assign bridges to users. Each bridge can accept as many participants as there are ports available for that group.

Bridge moderators (Centrex users) have access to a bridge management portal that is integrated into the Centrex Service Application Server CommPilot Personal portal. The portal allows for creating and managing bridges prior to the conference as well as moderating a live conference by adding, removing, muting, holding and retrieving participants.

The conferencing application is integrated with the user's personal information manager (PIM) so invitations to participants can be e-mailed to the moderator's contacts. Participants can dial into the conference or click a link in the invitation.

The moderator has access to enhanced functions during the conference:

- Hand raising
- Roll call
- Recording
- Playback

The Conferencing service also allows for document sharing (Excel, Word, and PowerPoint documents).

Provisioning

The group administrator manages conference bridges through the CommPilot Group web portal[13].

A conference bridge has to be created by the CAS system through EMA, so that the Conference Numbers are stored both in the AS and the HSS.

The group administrator assigns it a name, a maximum number of ports and one or more moderators. Each moderator has access to the *Bridge* page to start new conferences.

After being allocated a certain number of ports for the group, the administrator can create as many ports to the conference bridges that are created, as long as the total number of ports allocated to these bridges does not exceed the maximum number of ports allocated to the group.

During the conference, the moderator can manage the conference and its participants through the *Bridge Provisioning* page.

3.5.7 Instant Group Call

The Instant Group Call (IGC) service allows a user to call a group of members, whereby the system alerts all members in the group. As the members answer, they are joined into a multi-way conference.

The Instant Group Call service allows an administrator to define a group composed of a list of member users. These members can be part of the same group or enterprise (specified by user name, extension or location code + extension) or can be external users (specified by a phone number or SIP URI).

Description

The IGC service is a group service assigned to a group by a system or service provider administrator. Once the service is assigned to the group, then a system, service provider, or group administrator creates an instance of the service and sets its attributes.

Provisioning

Each service instance (also referred to as virtual user) is configured with the following:

- ID – The ID uniquely identifies the virtual user, for example, group1@domain.net.

[13] The virtual users to use as Conference numbers are provided by the Service Provider to the IMS network and the Application Server.

- Name – The name allows an administrator to provide a descriptive name that identifies the purpose of the virtual user, for example, “sales”.
- Phone Number/Extension – An administrator can assign a phone number and/or an extension to the virtual user.
- Calling Line ID Names – The calling line ID name is used when the virtual user alerts group members.
- Department – The administrator may assign a department to the virtual user.
- Language – The language associated with the virtual user.
- Time Zone – The time zone in which in the virtual user is defined.
- Aliases – The virtual user can be assigned up to three SIP-URI aliases.
- Enable Answer Timeout – A flag indicates if there is a maximum call time for unanswered calls.
- Maximum Call Time – The maximum call time for unanswered calls.
- List of Members – This is the list of members that are alerted when the virtual user phone number or extension is dialed. Up to 20 members can be defined for a virtual user. A member consists of an address that is used to reach the member. The address can be a SIP-URI, a phone number, location code and extension, extension or E.164 number. With the exception of SIP-URI, the address can be prefixed with a FAC, providing that the corresponding service is assigned to the virtual user.

An IGC virtual user can be assigned user services and may be subject to group service behavior when it applies to users. The following services typically apply for virtual users (for example, Hunt Group, Instant Conferencing, Call Center) and also apply for an IGC virtual user:

- Selective Call Acceptance
- Selective Call Rejection
- Anonymous Call Rejection
- Call Forwarding Selective
- Call Forwarding Always

- Call Forwarding Busy
- Do Not Disturb
- Voice Messaging User
- Third Party Voice Mail
- Priority Alert
- Calling Line ID Blocking
- Diversion Inhibitor
- Custom Ringback
- Incoming Calling Plan

3.5.8 Music On Hold

This service allows an administrator to set up and maintain a media source (audio or video) that can be broadcasted to held parties in various scenarios (Call Park, Call Hold, and Call Centres).

Description

Music On Hold is a group service that allows the group administrator to set up an audio source (music, advertising) that can be broadcasted to held parties in various scenarios.

The Music On Hold service is made up of two components:

Media source component – The media source is provided by the group administrator by uploading media files on the system. These media files are played back to held parties of applicable services. The media file that gets played back is selected based on the available formats and the capabilities of the party's endpoint, that is, if a video file is available and the party supports video, the video file is played back. Otherwise, the audio file is played back.

Broadcast component – The broadcast component allows the group administrator to enable selected services to use Music On Hold so that parties held through these services are played back the configured media source.

The following services can be enabled to use the Music On Hold service:

- **Call Hold** –This includes calls held by activation of a flash service or via the Call Manager.
- **Call Park**
- **Call Waiting** – This includes calls held by activation of a flash service or via the Call Manager.
- **Consultation Hold** – This includes Call Transfer and Three-Way Calls.

When no Music On Hold is specified or if Music On Hold is turned off for one of these services, the held or queued party hears silence.

External Source for Music On Hold

The group administrator can configure the Music On Hold Service to make use of an external audio source.

The audio source is controlled by the enterprise and is typically located on the enterprise's premises. When configured, the Centrex Service Application Server connects held parties to the audio source using SIP. The external music source then automatically answers the SIP call and plays music. It is assumed that the external music source accepts multiple simultaneous connections.

NOTE: This method, to provide music on hold, only supports audio media.

Call Hold and Call Park

The Call Hold and Call Park services use the audio source specified through the group *Music On Hold* service page on the CommPilot Group web portal. The page also allows for selectively activating and deactivating Music On Hold for each service.

The music source can be a system-provided audio file, or a custom audio file selected by the group administrator and uploaded to the system.

The music files uploaded by the administrator must be μ-law .wav files that are a maximum of ten minutes in length. The maximum size of this file is 4.6875 MB.

Alternately, the group administrator can also select an external source for the Music On Hold service. In this case, the administrator can select the device providing the audio among the list of authorized devices for the group.

Users can activate and deactivate the music on hold for themselves via the CommPilot portal and via a feature access code.

Call Centre

Each call centre can define and use its own music source. As for Call Hold and Call Park, the music source for each call centre can be the default system one, or a custom music source uploaded by the administrator. The same format and duration requirements apply for the call centre Music on Hold.

Department Music On Hold

This functionality allow for a separate music on hold audio sources to be configured on a per-department basis.

The per-department audio source is optional, and departments without their own audio source make use of the group-defined audio source by default, as usual.

Once the group administrator allows a department to use its own audio source, this audio source can be configured by the department administrator.

3.5.9 Series Completion

The Series Completion service is used to create an ordered list of users, and when a call attempts to terminate on one of these users and finds a busy condition, the call overflows to the next user on the list, until a free user is found or the end of the list is reached.

<u>Description</u>

Unlike hunt groups that use separate phone numbers all calls trigger the hunting capability, Series Completion is initiated for any call terminating on a member of the series completion group. Hence, a series completion group can be viewed as a call forward busy chain among selected members of a group.

<u>Provisioning</u>

The administrator configures the Series Completion service through the CommPilot Group web portal. The Provisioning page allows the administrator to enter the ordered list of users, making up the series completion group.

3.6 Group Services, Call Leg independent

3.6.1 Attendant Console

This web-based service enables a user (for example, a receptionist) to monitor a configurable set of users in their business group.

Description

The Attendant Console service provides critical call detail and group member status for effective attendant call routing. Entirely web-based, this service surpasses legacy PBX consoles and seamlessly combines with Centrex Service applications, such as Auto Attendant, for enhanced attendant solutions.

Components and Functionality

Functionality is distributed across three components to optimize attendant performance. The components and associated functionality are as follows:

- The Attendant Console is used to:
  - View member's information: status, name, number, extension, department, email, mobile, pager, and title. The attendant can select which column to display and in which order.
  - View call details (remote name, number, and duration)
  - Click-to-dial/Transfer
  - Multi-character jump in list and filtering capabilities
- An IP phone is used to:
  - Answer, Transfer, Hold/Retrieve, Release (Some IP hones support multiple call appearances.)
- CommPilot Call Manager (optional) is used to:
  - Answer, Transfer, Hold/Retrieve, Release
  - Provide directory assistance, web pop-ups
  - Integrate with Outlook contacts, send calls to voice mail

Combining these components allows attendants to view call details and initiate transfers in a several ways to accommodate a variety of user preferences.

Web Interface

The Attendant Console service is entirely web-based for flexible deployment and Provisioning. Moreover, an intuitive layout of station functionality minimizes attendant training and support. Web-based Provisioning allows quick addition and/or deletion of group members.

Moreover, Attendant Console supports flexible staffing arrangements and can be rapidly assigned to specific users at their desks with limited disruption. With instant messaging, attendants can quickly confirm user status to ensure correct call routing and relate critical instructions.

Enhanced Attendant Solutions

The modularity and flexibility of Centrex applications allow the creation of enhanced attendant solutions. Solutions are available as part of a Centrex service offering as follows:

- Multiple attendants (primary/alternate, multi-shift attendants):
  - Attendants log in and log out as needed.
  - Since the application is web-based, there is no need to change desks.
  - Functions behind a call centre front end.
- Night service:
  - Apply Selective Call Forwarding, whereby after-hours calls are forwarded to specified contacts.
- Company voice mail:
  - Transfer calls to the company voice mail or a call centre.
- Integration with Auto Attendant:
  - Overflow/Busy/After Hours calls go to an auto attendant.
- Combination with Instant Messaging:
  - Attendant sends instant messages to group members for call routing instructions.
- Configurable web pop-up:
  - Pop-up screen with customer relationship management (CRM) system information, in-house database records, or other important caller details.

Several enhanced Attendant Console solutions are unique to the Centrex Service Application Server and can be deployed for improved productivity and customer service performance.

Provisioning

The group administrator assigns the Attendant Console service to selected users and also determines which users are monitored by a given Attendant Console.

The attendant can select the user information (columns) appearing in the main Attendant Console window.

3.6.2 Configurable Calling Line ID

This capability allows a group administrator to assign to users an alternate Calling Line ID name and number, which is delivered to the called party.

Description

This capability provides the group administrator with the following fields on the user profile page:

- *Caller ID First Name* – The user's first name as it should be delivered as part of the caller ID.

- *Caller ID Last Name* – The user's last name as it should be delivered as part of the caller ID.

- *Caller ID Number* – The user's number has it should be delivered as part of the caller ID.

These attributes are optional. When they are left blank, the user's actual name and numbers are used by default.

Users themselves cannot modify these attributes, but can see them in their profile.

Provisioning

Group administrators configure the users' caller ID attributes through the CommPilot Group portal. All caller ID attributes are optional.

The usage of the Configurable Calling Line ID is determined by a system parameter. This system parameter can be configured to:

- Enable the use of configurable CLID for all calls.

- Enable the use of configurable CLID for all calls except emergency calls.

- Enable the use of configurable CLID for emergency calls only.

- Disable the use of configurable CLID for all calls.

3.6.3 CommPilot Group Web Portal

The CommPilot Group web portal allows group administrators to provision and configure resources for a group.

<u>Description</u>

The CommPilot Group web portal provides the group administrator with a web interface that allows for viewing, provisioning and configuring the resources of a group.

The following Provisioning and provisioning areas are available through the CommPilot Group web portal:

- *Group* – Allows for viewing and configuring the existing group resources like the group profile, departments, voice portal, and so on It also allows for tunnelling down into the user profiles and services.

- *Resources* – Allows for creating and deleting group resources like conference ports, users, phone number, devices, user services, group services, and so on

- *Group services* – Allows for configuring the group services like the hunt groups, call centres, auto attendants, feature access codes, messaging, and so on

- *Account/authorization codes* – Allows for configuring the Authorization Codes and Account Codes services, provisioning the valid codes, and so on

- *Service scripts* – Allows for creating and loading call processing language (CPL) service scripts that apply to the entire group.

- *Calling plans* – Allows for defining calling plans for the group, department, and users.

The CommPilot Group web portal also allows for creating other group administrators, managing passwords, and configuring miscellaneous items related to the group.

The CommPilot Group web portal is authenticated for each administrator with a user ID and a password, and provides a secure connection to the Centrex Service Application Server.

Provisioning

The group administrator can access the CommPilot Group web portal from any standard web browser to perform the Provisioning and provisioning tasks listed above.

3.6.4 Configurable Extension Dialling

This service provides the ability to map directory numbers to unique extensions to allow abbreviated dialling between users of a group.

Description

The configurable extension dialling service allows the users of a group to call one another using abbreviated dialling.

The abbreviated digit string ranges from two to six digits in length, and is set to the last *n*[14] digits of the user's phone number by default. The group administrator can change the default extension to any other value that is not already in use by another member of the group.

Once assigned, users' extensions can be used for dialling and for more intra-group routing application that require a phone number (for example, call forwarding simultaneous ringing, speed dial, and so on)

Extensions can also be assigned to routable group services like hunt groups call centres, and auto attendants.

Provisioning

The group administrator configures the Configurable Extension Dialling service through the CommPilot Group web portal.

- The length of the extension is the same for all users in the group and is selected through the *Extension Dialling* Provisioning page.

- Each user and virtual user's extension is populated by default with the last *n* digits of the user's phone number. For users without phone numbers and for other cases where the default extension is not appropriate, the group administrator can change the default extension through each user and virtual user's Provisioning page.

The CommPilot Call Manager *Group* tab and the group auto attendant are automatically provisioned with the extensions of the users within their scope.

[14] The length *n* of the extension is configurable by the group administrator but should at least allow for accommodating the number of users in the group.

3.6.5 Configurable Feature Codes

The Centrex Service Application Server allows group administrators to select the feature access codes (FAC) used to activate, deactivate, and program various services.

<u>Description</u>

This capability allows administrators to configure the feature access codes used by members of the group to activate, deactivate, program and configure various services.

A feature access code is defined as a string of two to five digits and special characters that is associated with a Centrex service or function, which is dialled by the members of the group to interact with this service or function.

Feature access codes are configurable by the group administrator and are subject to the following rules:

- A feature access code can be two to five digits in length.
- Special prefix characters (A, B, C, D, *, #) can only be used for the first two digits.
- The last digit must be number from 0 to 9.

Feature access codes must be unique within a group. It is possible to configure an alternate feature access code for each feature access code-based service. When an alternate feature access code is defined for a service or function, it can be used instead of the primary feature access code to interact with the associated function or service.

When no feature access code is associated with a service or function, users of the group cannot interact with this service or function through their phone.

The Centrex Service Application Server validates all the feature access codes to ensure that:

- There are no collisions between feature access codes.
- There are no collisions between feature access codes and extensions.

A second test is also applied when adding or modifying a user's extension. If a collision occurs when either a feature access code or an extension is modified, an error message appears. Also, when the length of extensions is increased, a warning appears stating that feature access codes and extensions may collide.

When there is a collision between a feature access code and an extension or a Speed Dial Code, the feature access code has precedence.

Provisioning

Feature access codes are configured by the group administrator through the CommPilot Group portal.

3.6.6 Department Administration Layer

Description

This feature creates a new department administrative layer to improve the management of large or geographically distributed groups.

Department Administrators

Like groups, departments can be assigned an administrator who shares most of the group administrator's privileges for the users and services assigned to the group. Hence, the group administrator can delegate most of the day-to-day service management of the department to a department administrator.

Specifically, department administrators have the following management capabilities:

- Add, modify, and delete users in their department.
- Modify auto attendants, call centres and hunt groups and instant conference bridges belonging to the department.
- View group directory and modify group common phone list.

The group administrator defines the administrative scope of the department administrators by assigning users and services to specific departments. The department administrators can manage users and services belonging to their department the same way the group administrator can.

Furthermore, the department administrator can create new users in the department and assign them services the same way the group administrator can. The department administrator can also delete users in the department.

Department Provisioning

The following services can be assigned to a department:

- *Auto Attendant* – When an auto attendant is assigned to a department, the department administrator can configure it. Also, assigning auto attendant to a department allows the option to restrict the scope of name dialling to the users in the department. This is especially useful with large groups to limit the number of collisions between user names.

- *Call Centres* – When a call centre is assigned to a department, the department administrator can configure it and populate the list of agents with any user from the group.

- *Hunt Groups* - When a hunt group is assigned to a department, the department administrator can configure it and populate the list of agents with any user from the group.

- *Instant Conference* - When an instant conference bridge is assigned to a department, the department administrator can configure it.

Furthermore, the Centrex Service Application Server CommPilot web portal is enhanced to allow the administrator to bulk-provision services with entire departments. Hence, when populating the agents in a call centre, the administrator can select a whole department at once instead of selecting the users one by one, for instance. The following services are enhanced with department bulk provisioning:

- Call Centre
- Hunt Group
- Account and Authorization Codes
- Series Completion
- Call Capacity management
- Call Pickup
- Attendant Console

Finally, this service allows the group administrator to assign phone numbers (DID) to departments. Any phone number assigned to a department shows up as such in the list of phone numbers when assigning new phone numbers to users or group services.

<u>Provisioning</u>

Departments are created by a group administrator and only the group administrator can assign or modify users or services for a department.

Once users and services are assigned to specific departments, they can be configured and managed by the department administrators. Any user created by the department administrator is automatically assigned to the same department.

Note that the use of departments is optional. Department selection for users and services may be omitted (left as *None*) which leaves provisioning at the group level.

3.6.7 Resource Inventory Reporting

This service allows a group administrator to generate a report on the resources used in the group and in each department.

Description

This capability allows a group administrator to generate reports on the resources used in the group and in each department and can select the information to be reported. The report is generated dynamically when an administrator submits a request. The report is sent by e-mail to the account provided on the *Inventory Report* page as an ASCII attachment file and is displayed in separate window in ASCII comma-separated value (CSV) format.

The resources reported include:

- Phone numbers
- Devices
- Users and departments
- Services

Reports are displayed on a web page in CSV format, which can be exported easily to a spreadsheet for sorting and archiving.

Provisioning

To configure inventory reports, the group administrator provides the e-mail address where the report is to be sent, and checks or selects the options for report generation to be included in the report. The following options are offered:

- Users
- Services

- Devices
- Phone numbers
- Department

The group administrator submits the form by clicking the View button.

The report is generated dynamically and is sent in ASCII CSV format to the specified email address. Also a window pops up and displays the Resource Inventory report in ASCII CSV format.

3.7 Messaging Services

3.7.1 Send to VM Feature Access Code

The Send to Voice Mail feature access code functionality allows a user to transfer his/her remote party (or parties) to the voice mailbox of any user in his/her group, at anytime during a call, and without using the CommPilot Call Manager. The target mailbox may be the user's own mailbox. The transfer is done by entering a configurable feature access code (FAC) while the remote party is on hold.

Description

This capability can be used when a party is on hold (meaning consultation hold). Then, from a new consultative call, the user dials the Direct Voice Mail Transfer FAC in order to initiate the transfer. A first-time user is guided by announcements that explain how to transfer the held party to the user's mailbox, or to anyone else's mailbox. Experienced users have the possibility of dialing through, and to perform the transfer without waiting for the prompts.

With the Direct Voice Mail Transfer functionality, a user can achieve the same transfers to a voice mailbox as the CommPilot Call Manager. The only exception to this is that a user can only transfer to Voice Mail after answer.

3.7.2 Third-Party Voice Mail Support

Third-Party Voice Mail Support facilitates the support and integration of an external voice mail platform. This capability is required to deploy Centrex Service Application Server with a third-party voice mail platform while retaining the integration of voice mail with other Centrex Service Application Service services.

Description

This feature allows forwarding of busy and/or unanswered calls to an external voice messaging platform.

- The destination can be a phone number or a URL.
- The destination is configured at the group level.
- It allows configuration, at the user level, of the user-part of the Diversion Header.

- It has the least level of precedence, which means that call forwarding services and voice messaging services have precedence over it.

Message Deposit

Incoming calls that reach a busy or no-answer condition are redirected to the third-party messaging server configured for the group.

Users can also redirect an incoming call to a third-party messaging system through the **Send to VM** button of the CommPilot Call Manager.

The address of the mailbox where the caller is redirected is determined by the *Custom Mailbox ID* configured for the user.

Once the call is answered by the third-party messaging system, the call control is handed off to the messaging system for further processing.

Message Retrieval

Users of the Third-Party Voice Mail Support service who are configured to use the external voice mail platform are able to retrieve their voice messages by:

- Pressing the **Messaging** button on their phone (if available), OR
- Calling their own number from their phone.

In addition, users of the Third-Party Voice Mail Support service are always able to retrieve their voice messages by:

- Calling the external voice mail platform voice portal number, OR
- Calling the external voice mail platform voice portal SIP-URL.

In cases 1 and 2, Centrex Service Application Server redirects the call to the external voice mail platform, which allows the users to retrieve their messages through the applicable procedure.

Interaction with Centrex Service Application Server CommPilot Call Manager

For any member of a group who is using this service, the **Send To VM** button of the CommPilot Call Manager is always visible and enabled, and can be used to redirect an active call to the messaging system.

Clicking the **Send To VM** button handles the call as follows:

Content of “Enter Phone Number” input box on the CommPilot Call Manager: Empty
Outcome: The call is redirected to the user’s mailbox

Content of “Enter Phone Number” input box on the CommPilot Call Manager: Filled
Outcome: The call is sent to the third-party voice mail platform, if configured for the target user.

Interactions with CommPilot Express

CommPilot Express is available to Third-Party Voice Mail subscribers. The CommPilot Express configuration page is customized to reflect the dispositions available from an external voice mail system.

Incoming Message Waiting Indicator

Incoming message waiting indicators (MWIs) with the customMailboxId of a user are handled by the Third-Party Voice Mail service. In addition, the Centrex Application Server is enhanced to process incoming MWIs sent with aliases.

Feature Access Code

Using a set of feature access codes (FAC), it is possible to enable/disable sending calls to voice mail with options All (always), unanswered and busy. The number of rings before the call is considered unanswered is also possible to set using a FAC.

A user can also using a FAC reach its Voice portal or set his/her call forwarding destination to *62 instead of the voice portal number or extension.

<u>Provisioning</u>

Third-party voice mail support is composed of two levels of configuration:

At the group level:

- Set whether or not the service is enabled (active).

- Set the phone number or URL of the external voice mail platform.
This parameter is not visible to a user or group administrator. It is visible and can be modified only by service provider and system administrators.

At the user level:

- Set whether or not the service is enabled (active). By default, the service is inactive for assigned users, regardless of the group-level setting.
- Set whether or not busy calls are redirected to the external voice mail platform. The default is "true".

- Set whether or not unanswered calls are redirected to the external voice mail platform. The default is "true".

- Set the number of rings before considering a call as being unanswered.

- Set the custom mailbox ID to use (as diversion header) when redirecting calls to the external platform. This parameter is not visible to a user or group administrator. It is visible and can only be modified by service provider and system administrators. The customMailboxId can be a phone number, SIP URL, or "empty" (then the value used to fill the diversion header is the E164 phone number of this user, or the E164 number of the group).

Feature Access Codes Activation

Call forwarding to voice mail on busy condition is possible to enable/disable using the FAC 40, enable by *40XXXXXXXXX and disabled by #40XXXXXXXXX.

After the FAC is dialed, the system plays a confirmation announcement "Your Voice Mail service is now set to [not] answer calls when you are busy. Thank you." It then releases the call.

Call forwarding to voice mail for unanswered calls is possible to enable/disable using FAC 40, enable by *41XXXXXXXXX and disabled by #41XXXXXXXXX.

After the FAC is dialed, the system plays a confirmation announcement "Your Voice Mail service is now set to [not] answer calls when you do not answer. Thank you." It then releases the call.

Call forwarding always to voice mail is possible to enable/disable using FAC 21, enable by *21XXXXXXXXX and disabled by #21XXXXXXXXX.

After the FAC is dialed, the system plays a confirmation announcement "Your Voice Mail service is now set to [not] answer calls immediately. Thank you". It then releases the call.

It is possible for the user to change the number of rings required for No-Answer handling by dialing a FAC code from his/her phone.

1) From a plain phone:

- User dials *610.
- User listens to the prompt “Please enter the number of rings before the system applies No-Answer handling for your incoming calls”.
- User punches one digit (valid digits are 0, 2, 3, 4, 5, 6).
- User hears the announcement “Thank you. The system will apply No-Answer handling after… ‘n’ …rings for your incoming calls”.

2) From a SIP phone or call manager, the user can still use the FAC as stated above, or also:

- User dials *610*n#.
- User hears the announcement “Thank you. The system will apply No-Answer handling after… ‘n’ …rings for your incoming calls”.

When the value of *number of rings required for No-Answer handling* is set to “0”, the system will generate a No-Answer condition before the Voice Mail service sees the incoming call. As a result, the No-Answer handling will apply.

The FAC to reach the Voice portal as default is 62.

3.7.3 Voice Messaging – Personal

This service enables users to record messages from callers for calls that are not answered within a specified number of rings or for calls that receive a busy condition.

Description

The Centrex Service – Media Server provides all of the features of a traditional voice messaging solution, plus:

- Message delivery to any specified e-mail account.
- Message waiting notification delivered to the phone.
- Integration of the messaging capabilities with Centrex services (call back, transfer, CommPilot Express, escape to extension, voice portal, and so on).
- Integration of hybrid messaging systems within an enterprise.
- Administrator and user self-management through a web interface.

Further sub-sections provide more details on the capabilities of The Centrex Service – Media Server voice messaging, specifically:

- Message depositing
- Message storage
- Message retrieval
- Message waiting notification

Deposit

Incoming calls to the user are sent to voice mail upon reaching a busy or no-answer condition. The caller is then played a greeting. There can be different greetings for busy and no-answer conditions and all greetings can be partially or fully customized by the user:

- Default busy greeting
- Default busy greeting with name
- Custom busy greeting
- Default no-answer greeting
- Default no-answer greeting with name
- Custom no-answer greeting

The caller can then leave a message or press 0 to transfer to an attendant. The attendant is configurable by the user and can be any valid phone number. If the caller leaves a message, he/she has access to the following functions:

- Long message warning tone
- Set the message status to urgent and/or confidential

- Review the message and erase, record it again or deposit it

Users can also configure their voice mail service to serve other phones, such as a cell phone. With this capability, users can forward any phone to the CommPilot voice portal phone number and have calls be sent directly to their mailbox greeting. This functionality is referred to as Voice Messaging Aliasing.

Video Support

Centrex Application Server messaging allows for providing a video greeting to video-enabled callers, and also allows callers to leave video messages for the user.

If users are authorized, they can upload video greetings in addition to audio greetings. Depending on the network condition (busy/no-answer) and the codecs supported by the calling party, the appropriate greeting is selected and played back.

The caller can then leave a message as usual, in audio or video format. Video messages are stored as .mov attachments to e-mails (instead of .wav). Users are notified that a specific message is in video format. In this case, messages can be retrieved via e-mail through the voice portal, but in this case, only the audio portion is played back.

Storage

Voice messages are stored on standard e-mail servers (POP3, IMAP4, Microsoft Exchange Server) as .wav audio files attached to e-mails. The voice messages can be stored on a default mail server (provided by the service provider or corporate server), or the user may elect to have voice messages sent to a private account.

Retrieval

Users can retrieve their e-mails from their location, from a third-party location or from any standard e-mail client.

When retrieving e-mails from their location, users simply dial the CommPilot voice portal phone number (or extension). The system prompts the user for their passcode. After entering the passcode, the user is informed of the mailbox status (how many urgent, new, expired, and saved messages) and can review the messages through a menu. While reviewing the messages, users can play the envelope, jump to next or previous message, skip ahead, skip back, pause, repeat, erase, save, reply, call back, forward, compose and send to a user or a distribution list.

When retrieving e-mails from an e-mail client, the user simply configures the client to collect e-mail from the e-mail server where the messages are stored. Messages are retrieved as .wav attachments to e-mails and can be listened to with standard audio software. Messages received as e-mail can be manipulated like any other e-mail (stored, forwarded, replied to, and so on).

Message Waiting Notification

When the user receives new messages, they can be notified by standard message waiting indication mechanism. Users can also request a notification to be e-mailed to a specific location, like a cell phone, when a voice message is received.

Immediate Voice Mail

This service adds the possibility to select 0 (or "None") rings, meaning to immediately apply No-Answer processing.

When the number of rings before No-Answer processing applies is set to 0, and the called party is busy, the busy processing is applied. The only exception to this is for users using a SIP device, and when the phone is off-hook but the user is not yet involved in a call. In such a case, although the phone is off-hook, the No-Answer processing applies, because the Application Server is not aware that the called party is off-hook.

When a user with the Sequential Ring service enabled receives a call and the number of rings before No-Answer processing applies is set to 0, the base location is not rung, and the service proceeds to the next location.

Feature Access Code

Using a set of feature access codes (FAC), it is possible to enable/disable sending calls to voice mail with options All (always), unanswered and busy. The number of rings before the call is considered unanswered is also possible to set using a FAC.

A user can also using a FAC reach its Voice portal or set his/her call forwarding destination to *62 (default) instead of the voice portal number or extension.

<u>Provisioning</u>

A broad range of Provisioning options is available to the user. Through the CommPilot voice portal, the user can record new greetings and record a personalized name that is played as part of the default system greeting.

Through the CommPilot Personal web portal, the user configures:

- The greetings to be played.
- The server where messages are stored (personal or default mail server).
- The mode of retrieval of voice mails.
- The number of rings defining the no-answer condition.
- The alias to allow other phones to use the messaging service.
- The passcode to retrieve messages through the CommPilot voice portal.
- The activation and deactivation of message waiting indication on the phone.
- The activation and Provisioning of message waiting indication to an e-mail address.
- Distribution lists

The CommPilot Personal web portal also allows users to upload wav files from their computer, to be used as greetings or a personalized name.

Feature Access Code Activation

Call forwarding to voice mail on busy condition is possible to enable/disable using the FAC 40, enable by *40XXXXXXXXX and disabled by #40XXXXXXXXX.

After the FAC is dialed, the system plays a confirmation announcement "Your Voice Mail service is now set to [not] answer calls when you are busy. Thank you." It then releases the call.

Call forwarding to voice mail for unanswered calls is possible to enable/disable using FAC 40, enable by *41XXXXXXXXX and disabled by #41XXXXXXXXX.

After the FAC is dialed, the system plays a confirmation announcement "Your Voice Mail service is now set to [not] answer calls when you do not answer. Thank you." It then releases the call.

Call forwarding always to voice mail is possible to enable/disable using FAC 21, enable by *21XXXXXXXXX and disabled by #21XXXXXXXXX.

After the FAC is dialed, the system plays a confirmation announcement "Your Voice Mail service is now set to [not] answer calls immediately. Thank you". It then releases the call.

It is possible for the user to change the number of rings required for No-Answer handling by dialing a FAC code from his/her phone.

1) From a plain phone:

- User dials *610.
- User listens to the prompt "Please enter the number of rings before the system applies No-Answer handling for your incoming calls".
- User punches one digit (valid digits are 0, 2, 3, 4, 5, 6).
- User hears the announcement "Thank you. The system will apply No-Answer handling after… 'n' …rings for your incoming calls".

2) From a SIP phone or call manager, the user can still use the FAC as stated above, or also:

- User dials *610*n#.
- User hears the announcement "Thank you. The system will apply No-Answer handling after… 'n' …rings for your incoming calls".

When the value of *number of rings required for No-Answer handling* is set to "0", the system will generate a No-Answer condition before the Voice Mail service sees the incoming call. As a result, the No-Answer handling will apply.

The FAC to reach the Voice portal as default is 62.

3.7.4 Voice Messaging – Group

The Voice Messaging Group service allows the administrator to configure group-wide attributes for the voice mail service.

Description

The Voice Messaging Group service allows the administrator of the group to select attributes of the Voice Messaging service that apply to the whole group:

- *Message aging* – Allows the group administrator to set a maximum duration for the storage of saved messages.

- *Mail servers* – Allows the group administrator to specify a default POP3 mail server for the group.

- *Mailbox sizes* – Allows the group administrator to set a maximum mailbox size for the group.

- *User Mailbox Settings* – This feature is used by a group administrator to allow or disallow a user to configure their own POP3/IMAP server. While the administrator can always perform Provisioning changes on behalf of users, the user may or may not be able to do so.

Note that these attributes apply only to the users of the group using the default group mail server.

Provisioning

The voice messaging group attributes are configured through the CommPilot Group web portal.

3.7.5 Voice Messaging – Service Provider

The Voice Messaging Service Provider service allows the administrator to configure service provider-wide attributes for the voice mail service.

For customers providing wholesale service, they need to be able to support multiple service providers. The “From” field used when sending an e-mail for message deposit and message notification needs to be set based on the service provider. This feature allows the “From” field to be configurable on a service provider basis instead of a system basis.

Description

This service allows the “From” field to be defined at the service provider/enterprise level. The provisioning at the service provider/enterprise level is similar to the current provisioning supported at the system level.

Voice Messaging uses the “From” field defined at the service provider/enterprise level if one exists. Otherwise, it reverts to the one defined at system level.

3.7.6 Voice Portal

The Voice Portal provides an interactive voice response (IVR) application that can be called by members of the group from any phone to manage their services and voice mailbox, or to change their passcode.

The group administrator can also use the Voice Portal to record new greetings for a group's auto attendants.

The voice portal allows users to automatically log in to the voice portal if calling from their own phone or device. This is the user option *Auto-login to voice portal if calling from own phone*. When set to "yes", then when users call in to the voice portal from their own phone, they are not prompted for a passcode, and instead, immediately access the voice portal menu. When set to "no", then the existing functionality is used and users are prompted for their passcode.

Description

The Voice Portal provides a convenient way for users to manage their services from any phone. The Voice Portal allows the users to:

- Manage their voice mailbox (see *Voice Messaging – Personal*):
    - Retrieve messages
    - Compose, forward, or reply to messages
    - Change greetings
- Activate, deactivate and program their Call Forwarding Always service (see *Call Forwarding Always*)
- Select a CommPilot Express profile (see *CommPilot Express*)
- Record a personalized name for an auto attendant and standard voice mail greetings
- Modify passcode
- Record auto attendant greetings (group administrator only)
- Make an external call

To access the Voice Portal menu, users must dial the number of their group voice portal or using the assigned Feature Access Code. A user can access the voice portal by dialing *62, instead of the number or extension of his/her voice portal. The result is the same.

A user can also set his/her call forwarding destination to *62 instead of the voice portal number or extension. Again, the result is the same (the caller rolls to the called party's voice mailbox in the BroadWorks Voice Mail system).

Each user can enable or disable auto-login to the voice portal. When the voice portal auto-login option is disabled, the login behavior remains unchanged. When enabled, all scenarios where the system recognizes the calling user (and would usually prompt immediately for a password rather than for an ID), result in an automatic authentication, and the password collection phase is skipped. Examples of automatically logging in to the voice portal are as follows:

- Centrex Application Server users call the system voice portal number from their own phone.
- Centrex Application Server users call their group voice portal number from their own phone.
- Centrex Application Server users call themselves from their own phone (the entry point is the VMR main menu).

Upon connecting to the voice portal, users are optionally played a branding announcement, followed by a prompt for their number and passcode[15]. Upon successfully authentication, users are presented with the main menu that offers the options described earlier[16].

Users can then select the desired option from the main menu and navigate through the menus by pressing the corresponding DTMF keys on their phone.

All options offered by the Voice Portal service allow users to revert back to the main menu, so multiple options can be selected during the same session.

Voice Portal Wizard

The Voice Portal wizard is optionally assigned to groups and assists users the first time they log into the voice portal.

Upon logging in, users are guided through the following steps:

- Change passcode from the default one (or after an administrator has reset it)
- Record personalized name

When the Voice Portal wizard is active for a group, all users must go through the wizard before they can use the voice portal for the first time.

Passcode Rules

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[15] If users call their voice portal from their own phone or from a phone for which they define a Voice Messaging alias, they are only prompted to enter their passcode if they have activated the auto-login option.
[16] The voice portal presents only the options corresponding to the services assigned to the user. If a user is not subscribed to a service offered by the voice portal, the option is not offered as part of the menu.

This feature enhances Voice Portal security by providing a set of rules to minimize Voice Portal access by unauthorized parties.

A system level default Voice Portal passcode rule is defined. When the service provider/enterprise has Voice Messaging Group service authorized, the default passcode rule is applied. Only the system administrator can change the system default passcode rule.

Each service provider however, can override the system default passcode rules. This modified set of rules is then used as the default rules for the groups within the service provider/enterprise. The group has the rule applied when the Voice Messaging Group service is authorized.

The Voice Portal passcode rules can also be overridden for each group, and ultimately define the rules that apply to all users of the group.

The passcode rules are described below and apply each time users change their passcode.

Passcode Aging

When enabled, this rule starts a timer when the user changes his/her passcode. The change can be performed via the CommPilot web portal or the Voice Portal.

When the timer expires, users are requested to select a new passcode before they are granted access to their Voice Portal (via a wizard-like IVR). The user hears "Your passcode is expired; Please enter a new one now to get access to the Voice Portal. Please enter the new passcode, followed by # sign." announcement.

The new passcode can be selected via the CommPilot web portal or the Voice Portal.

The administrator can configure the duration of the timer.

Passcode Rules

Users have to select a passcode that follows the rules defined by the administrator; otherwise their new passcode is rejected, and they have to choose a new one.

The following rules are defined. Each rule can be enabled and disabled independently:

Passcode Length – By default, the length of the passcode must be between 4 and 8 digits. When enabled, this rule allows for setting these boundaries to other values.

Trivial Passcode – When enabled, this rule rejects passcodes that are considered trivial:
- Repeated digits (for example, 11111, or 22222)
- The user's own extension or phone number
- The user's own extension or phone number reversed

Repeated Passcode – When enabled, this rule rejects the passcode that is the same as the previous passcode or a reversal of the previous passcode. This only applies when the user logs in. It does not take effect when the administrator modifies the user's passcode.

Passcode Lockout

This feature locks out a user Voice Portal access after *N* unsuccessful log in attempts in a row. Upon locking out a user account, an e-mail is sent to the group administrator with the user ID, the time of the unsuccessful attempt, and the Caller ID of the party for the last unsuccessful attempt.

When locked out, a user Voice Portal account must be reset by the group administrator via the user's *Passcode Reset* page before it can be used again. If the user tries to log in when the account is locked out, he/she hears the message "Your Voice Portal access is locked out. Please contact your group administrator to reset the passcode. This operation can not be completed at this time. Please hang up and try again later." announcement.

The group administrator can configure the value of *N*.

Enterprise Voice Portal

The system administrator can set the scope of the Voice Portal for the whole system, between:

- Group level – In this mode, any voice portal in the system serves only users of the group it belongs to.

- System level – In this mode, any voice portal in the system has the capability of redirecting any caller to the voice portal of their own group. In other words, under this mode of operation, all voice portals in the system work in a cooperative mode.

- Service provider/enterprise level – In this mode, voice portals defined in groups of a same enterprise act cooperatively, by redirecting users of the same service provider/enterprise to the voice portal of their own group, but they do not cooperate with voice portals defined outside the scope of the service provider/enterprise.

User can be identified by their extension or by their (location code + extension). This affects log in to the voice portal, compose/forward/reply, and also extends to distribution lists. Note that for users who are not part of an enterprise, the voice portal only supports phone numbers as a user identifier.

The “Send to whole group” is optional. For the voice portal scope, for a system or service provider/enterprise, this option is unavailable and hidden at configuration, whereas when voice portal scope is group, the option is available (can be turned on/off) and is configurable at the group level.

Scope of Voice Portal

The scope of any voice portal depends on the following configuration values:

Voice Portal Scope at System Level: System
Voice Portal Scope at Enterprise or Service Provider Level: Not configurable
Behaviour: Voice Portal cooperates with any other Voice Portal in the system

Voice Portal Scope at System Level: Configurable at enterprise or service provider level
Voice Portal Scope at Enterprise or Service Provider Level: Enterprise or service provider
Behaviour: Voice Portal cooperates with any other Voice Portal that is defined in a group that belongs to the same enterprise.

Voice Portal Scope at System Level: Configurable at enterprise or service provider level
Voice Portal Scope at Enterprise or Service Provider Level: Group
Behavour: Voice Portal does not cooperate with any other Voice Portal in the system

When the voice portal scope is set to “system” at the system level, the configuration value for voice portal scope at the enterprise level remains unchanged; it is only shadowed (made invisible and inapplicable) by the fact that the system administrator has set it to “system”.

Log in with Location Code + Extension

The login to the voice portal supports (location code + extension) for users who are part of an enterprise. This does not affect the ability for a given user to login through a given voice portal. This ability to login using location code + extension applies only when done in the context of a same enterprise.

Location code + Extension to Identify Destinations of Message Compose and Forward, and Distribution Lists

When sending a composed message, or forwarding a received message, destinations used to be identified by either an extension or a national number. The option to identify them using (location code + extension) is added. This also applies to pre-configured destinations stored in distribution lists.

In either case, the destinations are to be reached (receive the composed or forwarded message) only if they belong to the applicable scope of voice portal (system, enterprise, or group). Again, this ability to log in using location code + extension, applies only when done in the context of a same enterprise.

Option to "Send to Entire Group" for Message Compose and Forward

The "Send to Entire Group" option of Message Compose and Forward is to remain disabled (and not configurable) where the voice portal scope is larger than the group. Otherwise it would become configurable. It is to be configured directly in the *configuration* page of the voice portal.

When disabled, the option is simply not voiced to a user of the voice portal, and pressing the matching key (typically 4, although now configurable at the system level) results in an error message ("this key is not valid"), followed by a replay of the options, similar to pressing any other invalid key.

Voice Portal Calling

This feature allows an authenticated user to originate calls from the Voice Portal. This feature is particularly useful for traveling users that already access the Voice Portal to retrieve voice messages and configure services. Traveling users typically access the Voice Portal using a toll-free number and this feature allows them to originate calls that eventually get charged against their account. For similar reasons, this feature can be useful for the employee working at home that needs to make long-distance or international calls on behalf of the company. Dialing in to the Voice Portal first allows the subsequent long distance call to be charged to the company instead of the user's home line.

Once the Voice Portal authenticates the user, the user makes calls as if they were originated from their normal location. This means that services such as OCP, account/auth code will apply on the outgoing calls made from the Voice Portal. This also means that accounting records will be generated against the user's account.

The user can make as many calls as desired. The user can either wait for the remote party to hang up, or hit an escape sequence to originate a new call from the Voice Portal.

Voice Portal Customization

This feature allows the system administrator to customize the prompts and the keys that can be used to navigate through the menus.

For each menu and submenu of the Voice Portal, the association of keys to actions (choices of each menu) is to be made configurable. Are excluded:

- Voice Mail Deposit Menu and submenus
- Voice Portal Admin Menu and submenus
- Voice Portal Wizard Menu and submenus

Only the system provider administrator is allowed to change the system-wide configuration of keys for all the Voice Portals in the system.

When choosing a key for a menu option, the system provides the list of valid free keys from which a key can be selected. Some key values may not be listed if the Application Server has these keys reserved for non-configurable purposes. For instance, in the "send to distribution list" menu, keys 0 to 9 are reserved as identifiers for distribution lists. Therefore, in order to avoid any clash, these values for keys cannot be selected for actions in this menu (for example, "Repeat menu" cannot be assigned key 3, only * or #).

The key is either one digit (0-9), *, or # (or "none" when choosing to disable an optional menu option), except for the prompt to initiate a new call when using VP calling (currently set to "##"), in which case the selection is made of a sequence of two to three keys, where the inter-digit timeout cannot be configured (set to one second).

The concept of "any key", "remaining keys", or "choice between x keys" is not supported for a menu option. For example, a menu action cannot be configured as being triggered by any keypad key (0-9, *, #), any key not used in a menu (1, 4, 9, and #, assuming these are not yet assigned to any other option in the menu), or a set of keys ("Repeat menu" is * or #).

Prompts

The system introduces new announcements, with one announcement per menu option, and one announcement per key value. So for most languages, prompts are automatically constructed to list the options and their matching keys. For languages that do not follow the "For this menu action, …" + "… press 5" way of building sentences, a change in the customization of the VP keys also requires a re-recording of these new announcements.

This means that the Application Server builds a menu option prompt by playing the new announcement introduced by this feature and by automatically appending the appropriate announcement that voices the key number (by appropriate, we mean here the key that maps to the menu option as configured on the new web page).

This feature does not modify the order in which menu options are voiced, regardless of the key configuration chosen, although optional menu options not selected, are skipped.

Extended Availability of Options

With this feature, some sub-menu options are now offered from more than one menu. For example, the option "reply to a voice message" is now offered from the "Play Messages" menu, and from the "Message Handling Options" menu. This way, it is possible to customize the menus such that it is possible to reach the "reply" option directly from the "Play Messages" menu only, from the "Message Handling Options" menu only, from both menus, or from none.

Not all options are available from any menu, only the relevant options are presented in each menu. Without allowing complete re-organization and dynamic definitions of new menus, this allows for some flexibility on how menus are structured.

Provisioning

The group administrator configures the voice portal through the CommPilot Group web portal. The following parameters can be configured:

- Activate/deactivate the voice portal
- Assign a phone number to the voice portal[17]

[17] The virtual users to use as Voice Portal numbers are provided by the Service Provider to the IMS network and the Application Server.

- Assign an extension to the voice portal
- Assign an administrator passcode to the voice portal
- Activate/deactivate the Voice Portal Wizard
- Activate/deactivate voice portal log in using phone numbers or Voice Messaging Aliases (in addition to an extension)

The Voice Portal is service-aware, and only offers menu options that are available via the user services.

Furthermore, the following configuration applies to voice portal customization:

For each menu, the mapping between the available options and their assigned keys: option ↔ key option: any option available under a given menu (For example, change a greeting, enter a submenu, return to previous menu, and so on.) key: 1, 2, … ,9, 0, *, #

In a given menu, each option is defined as being either mandatory or optional. A mandatory option has to be assigned a key. An optional option can be assigned a key, but it can also have no key associated to it.

<u>Residential Voice Portal</u>

This feature is used to create a voice portal that spans all groups in a service provider without requiring a public phone number for each group voice portal. In addition, a user can be configured to use the service provider voice portal or the group voice portal. If a carrier is using the service provider voice portal, a user is assigned a service provider voice mailbox, which is unique for that service provider.

Description

Residential deployments are frequently implemented using one group per household and one user per occupant. All these groups are usually under a single service provider. Centrex Application Server supports a voice portal with a “service provider” scope. Each group has one voice portal, but when its scope is set to “service provider”, a user can dial any of the group voice portals to log in. The user is then silently redirected to his/her actual group voice portal.

So far, this redirection required a full national number (DN), thus consuming one additional DN per group (that is, per household in residential deployments). This enhancement removes this requirement. The redirection is done by using the Centrex Application Server user ID of the voice portal.

Provisioning

No configuration is required because the user ID is automatically created when assigning the Voice Messaging Group feature to a group.

3.8 Service Provider and Enterprise Services

3.8.1 Business Trunking

Business Trunking is a service that provides a new framework and introduces the concept of a “Trunk Group. The trunk group allows for licensing a maximum number of simultaneous calls that can be handled by a selected group of users, referred to as trunk users.

Business Trunk users are normal registered user in IMT. This solution can be integrated wit any CPE capable of registering users and complies with Registration Function Specification of IMT, Ref. [4].

<u>Description</u>

This framework enhances the Centrex Application Server data model to introduce the concept of a “Trunk Group” that shares the following characteristics.

The trunk group can pull some services from the Centrex Application Server such as:

- *Account/Authorization codes*
- *Calling Plans*
- *Auto Attendant*
- *Hunt Groups*
- *Call Centers*
- *Series Completion*
- *Conferencing*
- *Messaging*

Deployment Model

In a typical deployment, a group/site hosts zero or more trunk groups serving a large CPE, and zero or more regular users. In this model, user resources are authorized as usual to the group. In addition, one or more trunk group(s) is assigned to the group with the following attributes:

Trunk Group Name – This is name for the trunk group, which is unique within the group.

Call Capacity – This is the maximum number of concurrent calls that can be processed by trunk users in a trunk group. Call capacity can also be configured for incoming and outgoing calls. When the maximum number of calls is reached for a trunk group, or for incoming and outgoing capacities, the call is released with a BUSY release cause for a blocked termination and a FORBIDDEN release for a blocked origination.

List of Trunk Users – All calls processed for these users are accounted for by the trunk group call capacity. The CPE can serve more users than trunk users by having a auto-attendant/receptionist that routes the calls internally to them but from IMT system point of view these users are not known.

Trunking Device – This device is the access device that represents the CPE. There can only be one trunking device per trunk group, so all trunk users make use of that trunking device. The trunking device used in IMT should be smart proxy device type.

Provisioning

Call Capacity Allocation

The system provider allocates trunking calls to a service provider and the service provider then distributes those calls to their groups. The service provider and groups have a read-only view of the calls allocated to them.

No license checking is performed during these allocations which is why unlimited is an option.

Trunk Group

The users in a trunk group are defined as ordinary users of the Centrex Application Server meaning that they have to be registered as ordinary subscribers in the IMT system.

When modifying the trunk group, the same attributes are shown in addition to a list of group users using the trunk group. If any users are currently using the trunk group, the device selected may not be changed.

User Profile

The trunk group is associated to the user as if it were a device in the profile page of the user. The group administrator selects the trunk group and fills in the line port that is displayed based on the rules of the device associated with the trunk group.

3.8.2 Call Processing Policies

This feature provides explicit control of certain Centrex Application Server’s call processing behavior. The policies are configurable in a hierarchal manner. The user policies have the highest precedence and can defer to the associated group policy. The group policies have the next highest precedence and can defer to the associated Service Provider/Enterprise policy. The Service Provider/Enterprise policies have the lowest precedence and are defaulted to the system-wide defaults set on the Application Server.

<u>Forced Use of Uncompressed Codec</u>

Fax machines and modems require the use of a clear channel and an uncompressed Codec. This feature allows an administrator to force a user’s device to use an uncompressed Codec on a service provider/enterprise, group, and/or user basis.

On a per call basis, the system selects the policy based on the level at which it is configured. The user-level policies have the highest precedence, followed by the group level, the service provider and enterprise levels, and finally the system level.

Therefore, the user policy is used if it is configured and enabled. If it is not, then the group policy is used (if it is configured and enabled). And if the group policy is not used, the service provider/enterprise policy is used (again, if it is configured and enabled).

If none of these policies are enabled, the policies are defaulted to the system-wide defaults sent on the Application Server, the SDP is not manipulated, and the user is able to use any codec supported by the user's device.

<u>Calling Line Identity Restrictions</u>

This feature allows an administrator to specify if calling line identity restrictions are enabled or disabled for a group on a service provider/enterprise or group basis. When incoming calls are received, this feature is checked to determine if calling line identity restrictions are enforced or not for the group.

This feature also allows an administrator to specify if calling line identity restrictions are enabled or disabled for an enterprise on a service provider/enterprise basis. When incoming calls are received, this feature is checked to determine if calling line identity restrictions are enforced or not for the enterprise.

The group level policy for the group capability is used if it is configured and enabled. If it is not, the service provider/enterprise level policy for the group capability is used (if it is configured and enabled). If neither one of these policies is enabled, the Calling Line Identity restrictions for the group are not enforced for group calls for the user.

Maximum Number of Simultaneous Calls

This feature allows an administrator to specify the maximum number of simultaneous calls supported on a service provider/enterprise, group, and/or user basis. If a user exceeds the maximum number of simultaneous calls allowed on terminations, the calling party is provided Busy treatment. Busy treatment may include Voice Messaging, Call Forwarding Busy, and any other active service assigned to the subscriber which is invoked on Busy. If the terminating subscriber has no active services assigned which trigger on Busy, Busy tone is provided to the calling party. Note that this feature only counts originating and terminating calls for a subscriber. Redirected calls are not counted as part of this policy.

The user level policy for this capability is used if it is configured and enabled. If it is not, then the group level policy for this capability is used (if it is configured and enabled). And if the group policy is not used, the service provider/enterprise level policy for this capability is used (again, if it is configured and enabled). If none of these policies are enabled, the number of calls for the user is not limited.

Maximum Call Time for Answered Calls

This feature allows an administrator to specify the maximum call time, in minutes, for answered calls on a service provider/enterprise, group, and/or user basis. If a user exceeds the maximum time on a given call, the call is torn down and an abnormal release is indicated within the Call Data Record.

The user level policy for this capability is used if it is configured and enabled. If it is not, then the group level policy for this capability is used (if it is configured and enabled). And if the group policy is not used, the service provider/enterprise level policy is used (again, if it is configured and enabled). If none of these policies are enabled, the answered call time for a user is not limited.

Maximum Call Time for Unanswered Calls

This feature allows an administrator to specify the maximum call time in minutes for unanswered calls on a service provider/enterprise, group, and/or user basis. If a user exceeds the maximum time on a given call, the call is torn down and the calling party is given reorder tone.

The user level policy for this capability is used if it is configured and enabled. If it is not, the group level policy for this capability is used (if it is configured and enabled). And if the group policy is not used, the service provider/enterprise level policy is used (again, if it is configured and enabled). If none of these policies are enabled, the unanswered call time for a user is not limited.

3.8.3 Configurable Default Feature Access Codes

This feature adds a service provider level default feature access code (FAC) set. When new groups are created under that service provider, their FAC table is initialized based on the service provider’s defaults.

Description

Currently the Application Server supports a default system-wide Default feature access code setting. Any group that is created within the system has the same default FAC setting.

This feature removes the restriction for a default FAC prefix that applies to each group created in the system. Instead the Configurable Default feature access code setting allows service providers to be able to set the default FAC services. Service providers assign the default FAC prefix. And any group that is created under that service provider will be assigned the default FAC that was created by the service provider. This feature does not change how FAC works. The group administrator still has the privilege to change the assigned default FAC to any other allowed value.

The service provider administrator (or an administrator above), determines both the main and alternate default FAC for any group that is created within their scope.

When configuring the default FAC, the following rules apply:

- A FAC must be 2-5 digits in length.

- Special prefix characters [A, B, C, D, *, #] can only be in the first 2 digits.
- The last digit must be numeric [0-9].

When services are authorized to a group, a check is made to validate that::

- There are no collisions between FACs and extension numbers within the group.
- There are no collisions between FACs.

In case of collisions, the newly authorized FAC (to the group) is left blank and a warning message is generated in the Application Server log file.

<u>Provisioning</u>

The service provider can create default FAC values for various services. These FACs are assigned to groups as default FACs for authorized services.

The group *FAC Configuration* page provides a reset link used to reset the group FAC values back to the default value.

3.8.4 Large Enterprise Support

This activity introduces an Enterprise layer in the Centrex Application Server provisioning model. This layer allows customer to better model, administer, and manage large multi-site enterprises.

This layer resides in parallel with service providers. Thus, a system administrator can create service providers and/or enterprises. Each is administered separately.

Users in a Large Enterprise by using location code and extension numbers can reach each other. These users are normal registered users in IMT.

<u>Enterprise</u>

The concept of Enterprises is added to the Application Server. An enterprise should be used when a company has multiple sites or heavily geographically distributed users.

The Application Server models an Enterprise as a specific type of service provider. All the capabilities of the service providers are available to the Enterprise. A system provider now sees a list of service providers and a list of enterprises.

The existing service provider layer remains unchanged.

The web interface is enhanced to exhibit the enterprise at the enterprise, sites, and user levels.

Enterprise Creation Wizard

A new wizard is provided to assist the administrator in creating a new enterprise on the Application Server.

The first step is the creation of the enterprise itself along with the basic enterprise profile.

The second step is to add administrator(s) to the enterprise. These administrators carry out the daily management tasks related to the enterprise. One or more administrators can be created in this step.

The next step is to authorize the services that can be further assigned to the sites making up the enterprise.

The last step is to authorize phone numbers and blocks of phone numbers to the enterprise. These phone numbers can be further assigned to the sites and then to the users.

Enterprise Private Dialling

This capability allows for creating a private dialling plan shared between the multiple sites making up an enterprise on the Application Server. The enterprise dialling plan allows users of the enterprise to call one another using location codes and extensions instead of full phone numbers.

When creating the sites (group), the administrator can assign location codes to each group that will be used by enterprise users to make calls between sites, using a private dialling plan.

The enterprise calls using location codes and extension are as normal routed via CSCF as Distributed Group Calls (DGC).

Policies for Service Provider and Enterprise Administrators

Policies are added to the service provider (enterprise) administrators in a similar way to how they exist for group administrators today. The following policies are allowed:

Enterprise Profile Access:

- Full access to modify an enterprise’s profile. This is the default, which is the same as it is today.

- Read-only access to an enterprise’s profile. The *Enterprise: Profile - Profile* page is read-only.

- No access to enterprise's profile. The *Enterprise: Profile - Profile* is not shown on the Profile menu.

Group Access:

Full access to groups. This is the default, which is the same as it is today.

Restricted from adding or removing groups:

- The Add and Add Enterprise Wizard buttons on the *Enterprise: Profile - Groups* page are not shown.

- The Delete button on the *Group: Profile - Profile* is not shown.

No access to groups. The *Enterprise: Profile - Groups* is not shown on the Profile menu.

Administrator Access:

- **F**ull access to add, modify, and delete enterprise administrators. This is the default, which is the same as it is today.

- Read-only access to enterprise administrators. The *Enterprise: Profile - Administrators* is a read-only list.

- No access to enterprise administrators. The *Enterprise: Profile - Administrators* is not shown on the Profile menu.

Domains:

- Assignable only – Domains can be assigned to groups and users, but new domains cannot be added and existing ones cannot be deleted.

- Read and write (default) – Domains can be viewed, added, deleted, and modified.

Phone Number Access:

- Full access to phone numbers. This is the default, which is the same as it is today.

- Read-only access to phone numbers:

  - *Group: Resources - Assign Numbers is not shown on the resources menu.*

  - *All number assignments are read-only. The administrator cannot assign a number to a user or group service.*

Service Access:

- Full access to assigning resources to the group or users. This is the default, which is the same as it is today.

- Read-only access to service assignments:

  - *Group: Resources - Services is read-only, which is the same as the group administrator's view.*

  - *Group assignment pages are not shown (Assign Group Services, New User Services Template, Existing User Services)*

  - *User assignment pages are not shown (Assign User Services)*

Web Branding Access:

Full access to web branding. This is the default, which is the same as it is today.

No access to web branding. The *Enterprise: Utilities - Web Branding* page is hidden.

Priority Alert

The *Centrex Application Server Priority Alerting* page currently has (within the "Calls From" section), a button that allows the selection of "Any external phone number".

If this option is selected, then all calls received from outside the group, use distinctive ringing – even if the caller is in the same enterprise. This behaviour is modified by this enhancement.

If "Any external phone number" is selected, then the user should not receive distinctive ringing for any call originated within its enterprise (even if it is in a different group).

Enterprise-wide Departments

Managing the users in very large enterprises is enhanced by placing the users into departments.

Departments may be created either at the enterprise level or within a particular group. Departments belong to either the enterprise or the group in which they were created. A hierarchy of departments is supported in such a way that a parent department can have multiple sub-departments. A department created within a group can extend an enterprise department or another department within the same group. A department created within an enterprise cannot extend departments created at the group level.

All the departments that belong to a group must have a unique name within that group. Likewise, all the departments created at the enterprise level must have a unique name within the enterprise. However, it is possible to have duplicate department names in different groups or a department at the enterprise level with the same name as a department at the group level.

Users created within a group may be assigned to any department created at the enterprise level or departments created within the same group. In this way, departments can span across multiple geographic locations.

There are *web* pages to create and modify departments at the enterprise level. The existing *web* pages for managing group departments are changed to allow parent departments. All the *web* pages that display departments are changed to display the full path name of the departments including the parent department(s). In the full path name of the department, each sub-department is delimited from its parent department with a "\" symbol. If a group level department does not extend another group level department, the system automatically suffixes the department name with the group name in parenthesis.

Departments can be created at the enterprise level or at the group level. A group administrator can extend the enterprise department hierarchy, but cannot create departments at the top-most enterprise level. Users can belong to any department within the enterprise hierarchy unless the department belongs exclusively to another group.

Groups within service providers can also have a hierarchy of departments. In this case, all the departments belong to the group, because there is no enterprise, and the system does not insert the group name into the full path name of the departments.

The CLI and OSS are enhanced so that any command that takes a department as a parameter now allows specifying a department defined within the enterprise or a group. In addition, the CLI and OSS responses that include department names now specify the full path name of the departments.

It is not possible to create department administrators for departments defined at the enterprise level.

<u>Enterprise-wide Group Services</u>

The Centrex Application Server Large Enterprise framework allows group services, group rules, and dialing rules to be shared across the groups within the enterprise. These services and rules can be broken down into the following categories:

Terminating Services - These services require the ability to route calls to users across the enterprise. They include:

- Call Centers
- Hunt Groups

These services can be configured with agents belonging to different groups in the enterprise.

Rules – These capabilities can be defined by an enterprise, and inherited by all groups, or they can be defined on a group basis. They include:

- Digit Collection rules
- Extension Dialing rules
- Feature access codes
- LDAP configuration
- Password rules
- Voice Portal branding

Dialing – These functions allow a user to access other users by dialing their extension (when in the same site) or their location code plus extension (when not in the same site). They include:

- Auto Attendant
- Intercom
- Voice Messaging
- Voice Portal

These enhancements allow these services to be used between users belonging to different groups in the enterprise.

Enterprise Voice Portal and Messaging – For more information, see section Enterprise Voice Portal.

<u>Enterprise Directories and Usability</u>

Enterprise Directories

The enterprise directory is a list of all the assigned phone numbers in the enterprise. It includes users, Auto Attendants, Hunt Groups, Call Centers, instant conferences, and voice portal numbers. Each entry in the directory contains the name of the entity with their DN, extension, group, and department. There are new *web* pages at the enterprise, group, and user levels to view and search the enterprise directory.

There are also new web screens to allow the enterprise administrator to define a list of common phone numbers for the enterprise, similar to the existing group common phone list. The enterprise directory also shows the common phone numbers.

Users with the CommPilot Call Manager (CPCM) feature can view the enterprise directory from the CPCM. The CPCM company directory will continue to show the group directory for users within a service provider. The directory will show the enterprise directory for users within an enterprise.

Usability Enhancements

Many of the existing *web* pages are redesigned to improve performance or ease of use when managing very large numbers of users, DNs, or devices. The changes are at all levels, that is, system, service provider, group, and user, not necessarily enterprise-related.

The most common change involves presenting a search form prior to displaying a list of search results.

The search form allows you to search by multiple search criteria. Clicking the **plus** button adds a new search line. Clicking the **minus** button removes the last search line. The search can be as detailed as required. Clicking the **search** button finds the items matching all the specified search criteria.

3.8.5 Restricted Administrative Access

This feature adds granularity to the authorization of each administrative level of the CommPilot web portal. Currently, Centrex Service Application Server supports a multi-tier authorization scheme including: system administrator, provisioning administrator, service provider, group administrator, department administrator, and user. With this new feature, when a new administrator or user is provisioned, it is possible to set their specific level of control. For example, a group administrator can be created without the ability to add or remove users.

The administrative and user privileges on Centrex Service Application Server are currently fixed to allow and disallow certain actions. With this feature, when new users or administrators are created, “access rights” dictate which privileges they have on their *web portal* pages. Access rights can be qualified as “read-only” or “read and write”. Read-only access control makes the function available for viewing, but not for modification. Read and write access control makes the function available for both viewing and modification.

System Level Policies - Password Rules

System-level password rules include an option called “System and provisioning administrators; all other administrators and users use external authentication”. This option assumes that administrators and users are maintained outside of the system and the following web pages are hidden or modified:

Add Administrator (service provider) – Hidden.

Modify Administrator (service provider) – Change password is removed.

Add Administrator (group) – Hidden.

Modify Administrator (group) – Change password is removed.

Password Rules (service provider) – Hidden.

Password Rules (group) – Hidden.

Change Password (department) – Hidden.

Add User – Hidden (button is not shown).

Passwords (user) – Web access is removed.

System and provisioning administrators are not affected.

When the option, “System and provisioning administrators; all other administrators and users use external authentication” is chosen, the step to add an administrator is skipped for the *Add Service Provider* and *Add Group* wizards. The step numbers for subsequent pages are updated accordingly.

Provisioning Administration Level Policies

Read-only System or Provisioning Administrators

Read-only administrators at the system level view all the web pages as they do today, but are supplied with a modified read-only form that contains a **Cancel** button only.

Group Level Policies

Group policies are set by a service provider and determine what and how a group can administer itself. All administrators and users belonging to the group are restricted by these policies.

Extension Dialling

A group policy setting allows hiding the extension dialling page, making the extension dialling page *read-only,* or allowing *full* *access* as today. The extension page is always f*ull access* for service provider administrators.

LDAP Integration

A group policy setting allows the Light Weight Directory Access Protocol (LDAP) Directory web page to be hidden, making the LDAP directory page *read-only* or allowing *full access* as today. The *LDAP Directory* page is always *full access* for service provider administrators.

Department Administrator User Policy

A group policy setting controls department administrator access to users. The choices are:

- *Full access to users in that department.*
- *Restricted from adding or removing users; read-only user profile. This means the user profile is read-only.*
- *Restricted from adding or removing users; no access to user profile. This means the user profile is hidden.*
- *No access to users.*

User Authentication Policy

The user has a policy that determines if the *Authentication* page is available to them as it is today, is read-only, or it is hidden completely. This applies only to a user and not to an administrator.

User Profile Policy

The group profile setting determines if the *User Profile* page is available to them as it is today, is read only, or it is hidden completely. This applies only to users in the group and not to administrators.

Group Administrator Level Policies

Group administrator policies can be modified on the *Administrator Modify* page. When a group administrator is created, the default group administrator policy is applied. It controls access to users, departments, numbers, devices, services, administrators, and the group profile.

Group Profile Policy

The group administrator has a policy that determines if the *Group Profile* page is available to them as it is today, is read-only, or to hide it completely.

User Policy

The group administrator has a policy that is used to restrict access to users. There are four choices:

- *Full access to users.*

- *Restricted from adding or removing users; read-only user profile. This means the user profile is read only.*
- *Restricted from adding or removing users; no access to user profile. This means the user profile is hidden.*
- *No access to users.*

Department Policy (Full Access, Restricted, Not Available)

The group administrator has a policy that is used to hide the department web pages: *List*, *Add*, and *Modify*, to show only the department list, or operate with full access as is today. In addition, the assignment of departments to numbers is disabled unless the administrator has full access.

Numbers Policy (Full Access, Restricted)

The group administrator has a policy that is used to restrict them from performing number assignment. "Restricted" has the following changes:

- *The phone number for group services (Auto Attendant, Hunt Group, Call Center, Instant Conferencing, and Voice Messaging Group) is read-only for the administrator. The administrator is able to add these services, but not able to configure a number.*
- *The phone number in the add user and user profile page is read-only.*
- *The phone number in the alternate numbers page is read-only.*

Device Policy (Full Access, Restricted)

The group administrator has a policy that is used to restrict them from performing device-related changes. "Restricted" has the following changes:

- *Resources – Devices is a read-only list (cannot add or modify).*
- *Add User or User Profile has a read-only device section.*

Service Access Policy (Full Access, Restricted, Not Available)

The group administrator has a policy that is used to hide, make read-only, or allow full access to the service configuration of the group and/or users. The following occurs on the different access levels:

- *Full Access* is the same as today.
- For *Read-only*:

- *Services* page is currently read-only; no changes have been made.
- Assign Group Services are hidden.
- Assign Existing Services are hidden.
- Assign New User Services are hidden.
- Assign User Services at the user level are hidden.

- Administrator Policy (Full Access, Read Only, Not Available)

The group administrator has a policy that is used to restrict them from seeing the administrator list, allowing them to view only the administrator list, but not add, modify, or delete the list, or having full access as is today.

Policy Defaults

When creating groups or group administrators, a system default policy, which can be pre-configured, is applied. The exception to the default policies is made when other group administrators create group administrators.

Policy Dependencies

Policies have dependencies such that an administrator with limited access cannot create a user or administrator that has more access than they have today. The following dependencies are therefore implemented:

Group Administrator Creating Group Administrator

When a group administrator creates another group administrator, the policy of the new group administrator is set to the policy of the creating group administrator. The buttons, which give access to privileges greater than the creating administrator, are disabled.

Group Administrator Modifying Group Administrator

When a group administrator modifies another group administrator, the following buttons are disabled:

- All buttons for a policy that the group administrator being edited has greater privileges than the administrator performing the editing.
- All buttons, for a policy, which are greater than the group administrator performing the editing.

Provisioning

See above.

3.8.6 Service Packs

This activity introduces the concept of packages of user services that can be authorized and assigned as **a package of services rather than individual services.**

Description

The Service Packs feature allows service providers to group services together as atomic units that can be authorized and assigned according to their marketing strategy. Service packs are managed by service providers and do not impact how system providers authorize services to service providers.

Service packs consist of a name, a description that is visible to group administrators, one-to-many user services, and a quantity of packs. The quantity is the maximum number of a particular service pack that can be deployed to groups. Service packs consume the quantities of individual services given by the system administrator as soon as they are created.

Provisioning

Service packs are designed to support different variations of how a service provider packages services. The following service packs configurations are supported:

1) *Exclusive packs* – For this configuration, a service provider creates multiple packs with exclusive services. No two packs include the same services. A low-end user would have a single package of limited services and a high-end user would have multiple packages with each package adding additional services.
2) *Comprehensive packs* – For this configuration, a service provider creates multiple packs with duplicate services. In this strategy a user would always have a single pack. A low-end user would have a low-end pack and a high-end user would have a high-end pack.
3) *Combination packs* – For this configuration, a service provider creates multiple Exclusive User Packs and Comprehensive User Packs. Services in Exclusive User Packs would not exist in any Comprehensive User Packs. This strategy is provided to extend Comprehensive User Packs for "special" services.
4) *Unrestricted packs* – This package combination is supported for upgrading users from an "old" marketing strategy to a "new" marketing strategy. However, it is NOT recommended as a general marketing strategy because it is difficult to maintain.

This strategy consists of packages that contain the same services that can be assigned to the same user. For example, package A contains service 1, 2, and 3 and package B contains services 2, 3, and 4. When a user is assigned these two packages, they receive the superset of the services, in this case services 1, 2, 3, and 4. The reason to do something like this is to migrate from package A to package B without users losing their configuration data. In this case a user begins with package A. The service provider then adds the new package - package B. The service provider can then remove package A. The user ends up with services 2, 3, and 4 with services 2 and 3 remaining in their *original* configuration. If package A was removed and then package B was added, the user would have services 2, 3, and 4 with all the services in the *default* configuration.

Once a service pack is created, services cannot be added or removed. The name, description, and quantities can be modified. All service packs can also be made active or inactive which allows or prevents these packs from being sold to groups that do not yet have these packs authorized. This allows a service provider to deprecate old marketing schemes.

Service providers authorize and un-authorize a desired quantity of service packs to the group. A service provider can assign as many user packs as desired to a group, including ‘unlimited’. They can remove service packs unless service packs are assigned to a user.

Group administrators can assign and un-assign user packs to and from users. The users’ available services are the superset of all the services in the packs assigned. When a pack is un-assigned, that service is no longer available to the user and the user loses any configuration data associated with it if the service is not assigned individually or in another pack.

Group administrators are not able to individually distribute services in a service pack. The service pack is treated as a unit and cannot be broken or redistributed.

Individual services can be authorized and assigned as they were before, without using service packs, as well as in addition to using service packs.

Service Pack Migration Tools

The Service Pack Migration feature is a powerful tool for Service Providers to assign and un-assign services and service packs to groups of users.

Service Packs are an improved provisioning approach, so existing customers will likely want to migrate their users to Service Packs as soon as possible. While the conversion could be done manually, the Service Pack Migration feature greatly simplifies the process.

The Service Pack Migration feature can bulk-convert large groups of users matching a specified configuration to any new desired configuration. It is possible to assign or remove any combination of services and service packs. Since the process can be time consuming and processor-intensive, the migration tasks should be scheduled for execution during the night when system load is lightest. All of this is possible through an intuitive web-based interface.

Customers who are already using Service Packs can benefit from the Service Pack Migration feature as well. If a Service Provider decides to repackage the services into different Service Packs, the Service Pack Migration feature can be used to migrate users from the old packs to the new packs.

4 Operational Conditions

The Operator can configure all Feature Access Codes.

5 Revision History

Revision	Description
Rev F	Based on Rev D which was updated with new Feature Access Codes. Rev F updated with Calling Plan, Being Forwarded/Transferred
Rev G	Based on Rev E, additional info regarding Business Trunkig and Large Enterprise. Updated with Calling Plan, Being Forwarded/Transferred ar minor updates of External Emergency Routing.
Rev H	Clarified Business Trunking service and Large Enterprise.
Rev J	Adapted for IMT 3.0 R9A.
Rev K	Added clarification to Remote Office
Rev L	The Copyright info was corrected.

6 Glossary

Acronyms, abbreviations, and terms used are listed in the Glossary of Terms, see reference [1].

7 References

[1] Terms and Definitions,
0033-HSC 113 03

[2] Internal System Description,
1551-HSC 113 03/4

[3] User Guide for User and Service Provisioning,
9/19817-HSC 113 03/4

[4] Registration
5/155 17-1/HSC 113 03