The receptionist then communicates with the person who should handle the call, and provides them with the extension against which the call is held. That person calls the extension and retrieves the call.

If the call is not answered within a specified amount of time, it is recalled to the receptionist's device.

Only one call can be held at a station at any given time.

Usually the parking station, that is, a user account with the Automatic Hold/Retrieve service assigned should not have any device associated with it due to the calling limitations. For example, calls cannot be made or received and services that put calls on hold cannot be invoked from such a device. The emergency and maintenance calls cannot go through and feature access codes (for example, Music-On-Hold activation or Call Pickup requests) are blocked. Emergency calls made from a device with the Automatic Hold/Retrieve service activated receive a busy signal.

CONFIGURATION

This is not a regular user service. It is typically assigned to dedicated parking stations, that is, to user accounts reserved for holding/retrieving calls.

The group administrator can activate or deactivate the service and set the recall timer for the parking station.

BUSY LAMP FIELD

The Busy Lamp Field (BLF) service allows a user with a SIP attendant console phone to monitor the hook status and remote party information of selected users via the busy lamp fields on the phone.

DESCRIPTION

This service allows SIP attendant console phones to manage the lamp and console displays for monitored users.

The user configures the list of up to 50 users from their group or enterprise that they want to monitor. The SIP attendant console phone subscribes to the list and receives notifications about the state of the monitored users. For each monitored user, the device shows whether the user is busy and whether they are on an incoming or outgoing call.

The user can also be notified about calls being parked/no longer parked against monitored users.

The order of monitored users corresponds to the line appearance order of the monitored user on the SIP attendant console phone. If the monitored user list is modified (by adding, removing, or moving members) when there is an active subscription to the list, the subscription is terminated. The phone should then re-subscribe to the list.

CONFIGURATION

The user creates a list of users to monitor and assigns a SIP-URI to the list. The SIP-URI address must be on a domain available to the user and must be unique within the system among all SIP-URIs assigned to monitoring lists for the Busy Lamp Field service.

The user also specifies whether they want to be notified about parking events for monitored users.

Note: There is no limit to the number of monitoring lists on which a user can be listed.

CALL FORWARDING ALWAYS

The Call Forwarding Always (CFA) service allows a user to redirect all incoming calls to another destination automatically.

DESCRIPTION

This service redirects all incoming calls to a specified destination unconditionally, that is, independently of whether the user's line is busy, idle, alerting, and so on.

The destination (phone number or SIP-URI) to forward calls to must be permitted by the user's calling plans. Numbers representing emergency, repair, or chargeable directory assistance destinations are not allowed as a redirection destination.

A ring splash is applied to the user's device each time a call is forwarded.

Clearspan supports multipath forwarding for all types of Call Forwarding services. Thus, there are no restrictions on the number of simultaneous forwarded calls.

It is possible for the Call Forwarding Always service to create a call loop. For example, consider the case that arises when Subscriber A has Call Forwarding Always activated and configured to forward all calls to Subscriber B, and Subscriber B calls Subscriber A.

To avoid such loop conditions, Clearspan performs verification to make sure that the current forwarding attempt does not create a loop, before allowing the forwarding to occur. Clearspan considers the forwarding as a loop if any of the following conditions is met (after performing the translations for the redirection destination):

- Clearspan detects that the redirection destination is bound to a known Clearspan user who is already involved in the current communication.

- Clearspan finds a match between the phone number of the forwarding destination and the phone number of the originator of the call being forwarded. Note that this can be detected only if both locations represent a phone number.

CONFIGURATION

The user can control the service via the web portal, via feature access codes, or from the user's supported SIP phone.

In addition, the user can configure Call Forwarding Always remotely from any location through the voice portal.

When the user enters the forwarding phone number, the system validates the phone number against the user's calling plans. If the number is not allowed, the user is presented with an audio treatment or an error message.

The user can also set the forwarding destination to the Voice Portal feature access code, which is *62 (default) instead of the phone number, to forward all calls to the user's voice mailbox in the Clearspan Voice Mail system.

Web Portal

The user can activate and deactivate the service through the web portal. When the service is activated, a valid forwarding phone number or Uniform Resource Locator (URL) must be entered. The user can also select whether a ring splash should be played when a call is forwarded.

Feature Access Codes

The user can activate, deactivate, or inquire about the service through feature access codes dialed from the user's device.

- To activate the service, the user dials *72 (default), optionally followed by a valid forwarding phone number. The system then plays a confirmation announcement and the user hangs up.

  If the user does not enter a phone number, the previously configured phone number is used.

- To deactivate the service, the user dials *73 (default). The system then plays a confirmation announcement and the user hangs up.
- To obtain the current status and destination of Call Forwarding Always, the user dials *21* (default). The status is *active* or *inactive* and the destination is *voice mail* or the *current forwarding number*.

SIP Phone

The user can use the buttons on their supported SIP phone to activate this service.

Remote Access

The Call Forwarding Always service can be activated, deactivated, and programmed through an interactive voice response interface from any phone.

The user accesses this service by calling their group voice portal from any phone. After authenticating themselves, the user is 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, the user may hang up, or continue to navigate through other options of the voice portal.

CALL FORWARDING ALWAYS SECONDARY

The Call Forwarding Always Secondary service allows the user to configure a secondary forwarding address without losing configuration data of the Call Forwarding Always service.

DESCRIPTION

The Call Forwarding Always Secondary service allows the user to redirect all calls to another number or SIP-URI. This is a simplified version of the Call Forwarding Always service. Unlike the Call Forwarding Always service, Call Forwarding Always Secondary cannot be configured via feature access codes or voice portal.

Call Forwarding Always has higher precedence than Call Forwarding Always Secondary. If Call Forwarding Always successfully forwards a call, Call Forwarding Always Secondary is not applied to the call.

CONFIGURATION

The service can be configured via the CommPilot web portal, Open Client Interface-Provisioning (OCI-P) interface, or Xsi-Actions. A third-party application may subscribe to an Xsi-Events package via the Xtended Services Interface to receive notifications when configuration of the service is changed.

Note: The Call Forwarding Always Secondary service cannot be activated/deactivated via Feature Access Codes, device feature synchronization, or the voice portal.

The following can be configured for the user (via the web portal):

Service activation/deactivation – On/Off option button to enable or disable the service.

Forwarding address – The address to which the calls are redirected. This can be a SIP URI or a phone number. This parameter is required when the service is active.

Ring splash – An optional provisioning flag to indicate whether a ring splash is applied on the user's primary location each time a call is forwarded.

CALL FORWARDING ALWAYS TO VOICE MAIL

The Call Forwarding Always to Voice Mail service allows a user to send incoming calls to voice mail regardless whether the user is busy.

DESCRIPTION

This service redirects all incoming calls to the user's voice mail unconditionally, that is, independently of whether the user's line is busy, idle, alerting, and so on.

Note that the Call Forwarding Always service has precedence over Call Forwarding Always to Voice Mail service.

CONFIGURATION

This service is automatically available when the user has a Voice Messaging service (Voice Messaging User or Third-Party Voice Mail Support).

The user activates or deactivates Call Forwarding Always to Voice Mail either through the web portal or by using feature access codes.

- From the web portal, the user enables or disables the service via an option on the *Voice Management* page.
- From their phone, the user dials *21 (default) to enable the service and #21 (default) to disable it.

  After the feature access code 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.

  These feature access codes enable or disable the service for both Clearspan Voice Messaging and Third-Party Voice Mail Support. In general, a user has only one of the two services assigned, and that service is updated with these feature access codes. Should a user have both Voice Mail services assigned, both services are updated on each use of one of these two feature access codes.

CALL FORWARDING BUSY

The Call Forwarding Busy service allows a user to redirect incoming calls to another destination when the user is busy.

DESCRIPTION

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

The destination (phone number or SIP-URI) to forward calls to must be permitted by the user's calling plans. Numbers representing emergency, repair, or chargeable directory assistance destinations are not allowed as a redirection destination.

Clearspan supports multipath forwarding for all types of Call Forwarding services. Thus, there are no restrictions on the number of simultaneous forwarded calls.

CONFIGURATION

The user can control the service via the web portal, via feature access codes, or from the user's supported SIP phone.

When the user enters the forwarding phone number, the system validates the phone number against the user's calling plans. If the number is not allowed, the user is presented with an audio treatment or an error message.

The user can also set the call forwarding destination to the Voice Portal feature access code, which is *62 (default) instead of the phone number, to forward the calls to their voice mailbox in the Clearspan Voice Mail system.

Web Portal

The user can activate and deactivate the service through the web portal. When the service is activated, a valid forwarding phone number or URL must be entered.

Feature Access Codes

The user can activate, deactivate, or inquire about the service through feature access codes dialed from the user's device.

- To activate the service, 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.

If the user does not enter a phone number, the phone number that was previously configured is used.

- To deactivate the service, the user dials *91 (default). The system then plays a confirmation announcement and the user hangs up.
- To obtain the current status and destination of Call Forwarding Busy, the user dials *67* (default). The status is *active* or *inactive* and the destination is *voice mail* or the *current forwarding number*.

SIP Phone

The user can use the buttons on their supported SIP phone to activate this service.

CALL FORWARDING BUSY TO VOICE MAIL

The Call Forwarding Busy to Voice Mail service allows a user to redirect incoming calls to voice mail when the user is busy.

DESCRIPTION

This service redirects incoming calls to the user's voice mail when the user is busy. A user is considered busy by this service when there are too many active calls or a service (for example, Do Not Disturb or Selective Call Rejection) makes the user appear busy to the caller.

The Call Forwarding Busy service has precedence over Call Forwarding Busy to Voice Mail.

CONFIGURATION

This service is automatically available when the user has a Voice Messaging service (Voice Messaging User or Third-Party Voice Mail Support).

The user activates or deactivates Call Forwarding Busy to Voice Mail either through the web portal or by using feature access codes.

- From the web portal, the user enables or disables the service via an option on the *Voice Management* page.
- From their phone, the user dials *40 (default) to enable the service and #40 (default) to disable it.

After the feature access code is dialed, the system plays the confirmation announcement, “Your Voice Mail service is now set to [not] answer calls when you are busy. Thank you.” It then releases the call.

These feature access codes enable or disable the service for both Clearspan Voice Messaging and Third-Party Voice Mail Support. In general, a user has only one of the two services assigned, and that service is updated with these feature access codes. Should a user have both Voice Mail services assigned, both services are updated on each use of one of these two feature access codes.

CALL FORWARDING NO ANSWER

The Call Forwarding No Answer service allows a user to redirect incoming calls to another destination when the user does not answer within a specified number of rings.

DESCRIPTION

This service forwards incoming calls to a specified destination when a user does not answer a call for a configured number of rings or when an applicable failure occurs.

Call failures that trigger Call Forwarding No Answer are failures that occur before the no answer timeout except for conditions handled by other services, for example, a busy condition detected by Call Forwarding Busy, lack of credits detected by Prepaid, forbidden call detected by Session Admission Control or a screening service (such as Selective Call Acceptance or Selective Call Rejection), or a release on Personal Assistant failure.

Note also that Call Forwarding Not Reachable has precedence over Call Forwarding No Answer so it will trigger first when applicable (that is only for call failures, and only when an 18x response has not been received if so configured).

The destination (phone number or SIP-URI) to forward calls to must be permitted by the user’s calling plans. Numbers representing emergency, repair, or chargeable directory assistance destinations are not allowed as a redirection destination.

Clearspan supports multipath forwarding for all types of Call Forwarding services. Thus, there are no restrictions on the number of simultaneous forwarded calls.

The user has the option to select “0” (or “None”) rings, to immediately apply no-answer processing.

However, 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, 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 Clearspan is not aware that the called party is off-hook.

CONFIGURATION

The user can control the service via the web portal, via feature access codes, or from the user's supported SIP phone.

When the user enters the forwarding phone number, the system validates the phone number against the user's calling plans. If the number is not allowed, the user is presented with an audio treatment or an error message.

The user can also set the call forwarding destination to the Voice Portal feature access code, which is *62 (default) instead of the phone number, to forward the calls to their voice mailbox in the Clearspan Voice Mail system.

It is strongly recommended to avoid setting the number of rings before the call is considered unanswered to a value that is equivalent to the value assigned to the Maximum Duration for Unanswered Calls call processing policy. The concurrency between the two timers can cause one or the other to trigger first based on the activity on the server and the session topology. Therefore, a spacing of at least one second is recommended. The number of rings can be converted into duration with the ring period based on the country code setting.

The user can activate and deactivate the service through the web portal. When the service is activated, a valid forwarding phone number or URL must be entered. The user can also configure the number of rings before the call is forwarded.

Feature Access Codes

The user can activate or deactivate the service, configure the number of rings, or inquire about the service through feature access codes dialed from the user's device.

- To activate the service, 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.

  If the user does not enter a phone number, the phone number that was configured previously is used.

- To deactivate the service, the user dials *93 (default). The system then plays a confirmation announcement and the user hangs up.
- To obtain the current status and destination of Call Forwarding No Answer, the user dials *61* (default). The status is *active* or *inactive* and the destination is *voice mail* or the *current forwarding number*.
- To configure the number of rings before the call is forwarded, the user dials *610 (default).

  Note however, that this setting applies to all services with no-answer handling, that is, the Voice Mail, Third-Party Voice Mail Support, and Sequential Ringing services.

SIP Phone

The user can use the buttons on their supported SIP phone to activate this service.

CALL FORWARDING NO ANSWER TO VOICE MAIL

The Call Forwarding No Answer to Voice Mail service allows a user to redirect incoming calls to voice mail when the user does not answer the phone.

DESCRIPTION

This service redirects incoming calls to the user's voice mail when the user does not answer the phone within a configurable number or rings.

The Call Forwarding No Answer service has precedence over Call Forwarding No Answer to Voice Mail.

CONFIGURATION

This service is automatically available when the user has a Voice Messaging service (Voice Messaging User or Third-Party Voice Mail Support).

The user enables or disables Call Forwarding No Answer to Voice Mail either through the web portal or by using feature access codes.

- From the web portal, the user enables or disables the service via an option on the *Voice Management* page.
- From their phone, the user dials *41 (default) to enable the service and #41 (default) to disable it.

  After the feature access code 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.

  These feature access codes enable or disable the service for both Clearspan Voice Messaging and Third-Party Voice Mail Support. In general, a user has only one of the two services assigned, and that service is updated with these feature access codes. Should a user have both Voice Messaging services assigned, both services are updated on each use of one of these two feature access codes.

CALL FORWARDING NOT REACHABLE

The Call Forwarding Not Reachable service allows users to redirect incoming calls to a configurable destination when the user's device is unreachable or unregistered.

DESCRIPTION

This service forwards incoming calls to a specified destination when the user's device is not accessible by Clearspan. The destination (phone number or SIP-URI) they forward their calls to, must be permitted by their Outgoing Calling Plan. Numbers representing emergency, repair, or chargeable directory assistance destinations are not allowed as a redirection destination.

Clearspan supports multipath forwarding for all types of Call Forwarding services. Thus, there are no restrictions on the number of simultaneous forwarded calls.

CONFIGURATION

The user can control the service through the web portal or via feature access codes.

When the user enters the forwarding phone number, the system validates the phone number against the user's calling plans. If the number is not allowed, the user is presented with an audio treatment or an error message.

Web Portal

The user can activate and deactivate the service through the web portal. When the service is activated, a valid destination phone number must be entered.

Feature Access Codes

The user can activate, deactivate, or inquire about the service through feature access codes dialed from the user's device.

- To activate the service, the user dials *94 (default), optionally followed by a valid forwarding phone number. The system then plays a confirmation announcement and the user hangs up.

  If the user does not enter a phone number, the phone number that was configured previously is used.

- To deactivate the service, the user dials *95 (default). The system then plays a confirmation announcement and the user hangs up.
- To obtain the current status and destination of Call Forwarding Not Reachable, the user dials *63* (default).

SIP Phone

The user can use the buttons on their supported SIP phone to activate this service.

CALL FORWARDING SELECTIVE

The Call Forwarding Selective service allows a user to redirect their incoming calls to a specified destination based on selective criteria specified by the user.

DESCRIPTION

Call Forwarding Selective forwards an incoming call to a specified destination when the call matches a set of selective criteria specified by the user.

The criteria are based on the incoming caller identity and the time of the call, and are combined into criteria entries, for example, incoming calls from this number, within business hours, and during the workweek. For more information, see Selective Criteria.

Multiple criteria entries can be defined and calls can be forwarded to a different destination depending on the criteria entry that they satisfy. The service cannot be activated unless there is at least one active criteria entry defined.

Optionally, a ring splash is applied to the user's device each time a call is forwarded.

Clearspan supports multipath forwarding for all types of Call Forwarding services. Thus, there are no restrictions on the number of simultaneous forwarded calls.

CONFIGURATION

The user can control the service through the web portal or via feature access codes.

When the user enters the forwarding phone number, the system validates the phone number against the user's calling plans. If the number is not allowed, the user is presented with an error message.

Web Portal

The user defines the criteria entries used to filter incoming calls, and associates the same or different forwarding destination with each criteria entry. In addition, criteria entries can be selectively activated or deactivated.

The user can also activate and deactivate the service. When the service is activated, a valid forwarding phone number or URL must be entered and at least one active criteria entry must be defined.

The user can also specify whether a ring splash should be played when a call is forwarded.

Feature Access Codes

The user can activate, deactivate, or inquire about the service through feature access codes dialed from the user's device.

- To activate the service, the user dials #76 (default). The system then plays a confirmation announcement and the user hangs up.
- To deactivate the service, the user dials #77 (default). The system then plays a confirmation announcement and the user hangs up.
- To obtain the status and destination of Call Forwarding Selective, the user dials *54* (default).

CALLING LINE ID BLOCKING OVERRIDE

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

In other words, the user receives the calling line identity if available, regardless whether the calling line identity is blocked. The user never receives a calling line identity indicating "private".

DESCRIPTION

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 (both name and number) to the user if it is available.

The caller information provided to the user is bound by what Clearspan receives from the calling party and from other peripheral systems. For instance, if the caller's name is blocked in the Caller ID with NAMe (CNAM) database and cannot be obtained by Clearspan, the user only gets the calling number.

When this service is active, all calling line ID-based services behave as if the calling line 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.
- Call logs show calling line IDs for all incoming calls.

CONFIGURATION

The Calling Line ID Blocking Override service requires the Calling Line ID Delivery service to be assigned to the user (and activated).

The user can activate or deactivate the Calling Line ID Blocking Override service.

CALLING LINE ID DELIVERY

This service is a terminating service that delivers the identity of the calling party to the user who has the service to the device (if capable).

DESCRIPTION

Calling Line ID Delivery relays a caller's identity to the user's device, if the device is capable of displaying such information. The identity of the caller is delivered for every call that terminates to the user. The identity includes the calling party's number and name, if available.

If an incoming call is redirected or blocked before it can terminate, or if the user is busy, the identity is not delivered.

The Calling Line ID Delivery service is divided into two features: External Calling Line ID Delivery and Internal Calling Line ID Delivery.

FEATURE	DESCRIPTION
External Calling Line ID Delivery Feature	This feature allows the user to view the calling line ID information of another user in a different group. This feature also applies to intragroup calls that use the Network Server.
Internal Calling Line Delivery Feature	This feature allows the user to view the calling line ID information of another user within the same group.

CONFIGURATION

The user can independently enable and disable the external and internal calling line ID delivery.

CALLING LINE ID DELIVERY BLOCKING

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

DESCRIPTION

Calling Line ID Delivery Blocking blocks the delivery of a user's identity (both name and number) to a called party.

Calls made by the user to parties outside of the group or enterprise have the presentation of their identity (name and number) blocked.

Calling Line ID Delivery Blocking Per Call

The user 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.

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

Calling Line ID Delivery Per Call

The user can override the persistent blocking of the calling line ID delivery to 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.

This service is the exact opposite of Calling Line ID Delivery Blocking Per Call (see the section– Calling Line ID Delivery Blocking Per Call).

If the user's identity is not persistently blocked, the service is used the same way but has no impact.

CONFIGURATION

Users can activate or deactivate the Calling Line ID Delivery Blocking service via the Personal web portal, or by dialing a configurable feature access code from their phone.

- To activate the service, the user dials *31 (default). The system plays a confirmation tone and the user hangs up.
- To deactivate the service, the user dials #31 (default). The system plays a confirmation tone and the user hangs up.
- To block the delivery of their identity for the next call, the user dials by *67 (default) from their device before making the call. The system plays a confirmation tone followed by a dial tone. The user can then make the call as usual and their identity is blocked.

- To override calling line ID blocking for one call, the user dials *65 (default) from their device before making the call. The system plays a confirmation tone followed by a dial tone. The user can then make the call as usual and their identity is delivered to the far end.
- To obtain the current status of Calling Line ID Delivery Blocking, the user dials *54* (default).

CALLING NAME DELIVERY AND CALLING NUMBER DELIVERY

The Calling Name Delivery and Calling Number Delivery services allow independent delivery of calling name and calling number to the user.

DESCRIPTION

Calling Name Delivery and Calling Number Delivery are separate services, which allow delivery of calling name and number to Clearspan users independently of each other.

Users with Calling Name Delivery assigned and enabled are presented with the calling name, and users with Calling Number Delivery assigned and enabled are presented with the calling number of incoming calls.

The services may be enabled or disabled separately for internal and external calls.

The Internal Calling Line ID Delivery and External Calling Line ID Delivery services have precedence over the Calling Name Delivery and Calling Number Delivery services. If either Internal Calling Line ID Delivery or External Calling Line ID Delivery is assigned to a user, then the assignment or configuration of either the Calling Name Delivery or Calling Number Delivery has no effect. For example, if a user has Internal Calling Line ID Delivery assigned and disabled and Calling Number Delivery assigned and enabled for internal and external calls, and the user receives an external call, the calling number is not delivered to the user. The Calling Number Delivery service has no effect because the user also has the Internal Calling Line ID Delivery service assigned.

Calling Name Delivery may be blocked, if the user receives an external call from a Clearspan user with the Block Calling Name for External Calls policy enabled:

- For a user who is part of a service provider, an external destination is a destination outside the user’s group.
- For a user who is part of an enterprise, an external destination is a destination outside the user’s enterprise.

CONFIGURATION

The user can enable and disable the Calling Name Delivery and Calling Number Delivery services separately for internal and external calls.

Calling Name Delivery blocking for external calls may be configured by administrators at the system, service provider, and group levels.

CALLING NAME RETRIEVAL

The Calling Name Retrieval service allows Clearspan to provide a caller's name to a user by retrieving the calling name from a Public Switched Telephone Network (PSTN)-hosted database through a Signaling System 7 (SS7)-enabled network element such as a softswitch.

DESCRIPTION

This service provides a SIP subscription-based method or a Simple Object Access Protocol/ Extensible Markup Language (SOAP/XML) query method of retrieving calling name information from an external database on a per-call basis; this function is analogous to the GR-1188 Transactional Capabilities Application Part (TCAP) terminating query.

The basic service works as follows:

- If the name information is already present in the incoming call setup message, then the external database is not accessed.
- If the name information is not available, Clearspan sends a request to the external database.

The query contains the caller's number, which allows the external database to look up the caller's name. When Clearspan receives a response from the external database, the caller's name information is extracted from the message and is relayed to the user.

In addition, the service can be configured to trigger a query (SIP or SOAP) whether or not the calling name information is provided in the incoming INVITE and when the caller is a Clearspan user known in the Calling Name Retrieval subscriber's Application Server.

CONFIGURATION

The Calling Name Retrieval service can be configured to use either the SIP subscription-based method or the SOAP/XML query method.

In addition, the following service aspects are configurable:

- Whether to query the database for all network calls
- Whether to query the database for group/enterprise calls
- Domain Name System (DNS) Service Locator (SRV) support flag
- External database information
- Query response timeout

- Redundancy settings (to allow querying more than one database)

The system administrator configures the service at the system level using the command line interface (CLI).

The user can activate or deactivate the service using the web portal.

CALLING PARTY CATEGORY

The Calling Party Category (CPC) service allows a category to be associated with a user and used to determine the policies to apply to the user's calls.

DESCRIPTION

An administrator assigns a calling category to a user, which is then included in the signaling for all outgoing calls and used to determine the policies to apply to the user's calls and to route the calls.

The calling category assigned to a user is associated with all originating, forwarded, and transferred calls from the user to any Public Switched Telephone Network (PSTN) party or to another Clearspan user with the same service in a different Clearspan group.

Depending on the trunking device Clearspan is interacting with for a given call, Clearspan automatically populates the user's calling category in the appropriate format, either the calling category or Integrated Services User Part (ISUP) Originating Line Information (OLI), so it can be passed to that trunking device.

The following table lists the default calling categories provided by Clearspan.

CATEGORY	DESCRIPTION
Ordinary	The user has no special characteristics.
Payphone	The user represents a smart pay phone.
Prison	The user is from a prison.
Hotel	The user is from a hotel or motel.
Hospital	The user is from a hospital.
Special	The user should always be routed to an operator service system.

During a call forward or call transfer to another user by the terminating party in a different group, and in cases in which the terminating party is assigned a category, the category of the terminating party is used to forward/transfer the call.

CONFIGURATION

The system administrator can configure calling categories relevant to the networks they are connecting to for passing to outside networks.

The administrator configures the calling category for the user. The user can only view this information.

In some countries, this is a regulatory requirement used to identify the carrier of choice for the user (similar to equal access in the United States). The category assigned to the user maps to an intercity or international carrier that is then selected for routing outside the Clearspan system.

CALL LOGS

The Call Logs services allow users to view information about their placed, received, and missed calls.

DESCRIPTION

Call logs for placed, received, and missed calls are automatically saved when call events occur.

Users can view the logs through the web portal. The call log information includes the name of the remote party, the phone number, and the time the call was made. When displayed through the web portal, call logs are available under three tabs, one for each call type (placed, received, or missed).

Basic Call Logs

A maximum of 20 of the most recent logs per call type (placed, received, and missed) are stored for a user. When the maximum number is reached and a new call log is added, the oldest log of the same type is deleted.

Call logs are stored locally on the Application Server.

CONFIGURATION

The Call Logs service has no configuration.

The enterprise administrator configures the Call Logs group web policy to allow users with the Call Logs service to delete their call logs through the web portal. The policy is set on a per-group basis.

CALL ME NOW

The Call Me Now service allows an external party to initiate a Click-To-Dial call to a Clearspan user. For example, a Clearspan user can place a Call-Me-Now link on a web site and let non-Clearspan parties request to be called by them. The call is established by Clearspan at no cost to the external party.

DESCRIPTION

When an external party initiates a Call-Me-Now request to a Clearspan user, they are prompted to provide a phone number where they want to be called back.

Clearspan then screens the provided number using the selective criteria set by the Clearspan user.

The criteria are based on the incoming caller identity and the time of the call, and are combined into criteria entries, for example, incoming calls from this number, within business hours, and during the workweek. For more information, see Selective Criteria.

If no selective criteria are defined or all criteria are inactive, all calls allowed by the user's calling plans are accepted.

If the screening is successful, Clearspan initiates a call to the provided number. From the external party's perspective, the call is a regular incoming call from the Clearspan user.

The external party answers the call, and if required by the Clearspan user's Call-Me-Now configuration, the party is prompted to confirm the call.

If the answer confirmation is successful, Clearspan initiates a call to the Clearspan user. From the Clearspan user's perspective, the call is a regular incoming call from the external party.

The Clearspan user answers the call, and is now in a conversation with the external party.

Note that since the Clearspan user's terminating services apply to the call, it is possible that the call is redirected or given treatment instead of being answered by the user.

CONFIGURATION

The user can define and selectively activate/deactivate criteria entries used to screen phone numbers provided by external parties requesting to be called.

The user can also configure the service to request answer confirmation from the external party.

CALL NOTIFY

The Call Notify service allows a user to receive e-mail notifications about selected incoming calls based on the specified selective criteria.

DESCRIPTION

The user can define selective criteria that cause certain incoming calls to trigger an e-mail notification that contains information about the caller to be sent to the address specified by the user.

The criteria are based on the incoming caller identity and the time of the call, and are combined into criteria entries, for example, incoming calls from this number, within business hours, and during the workweek. For more information, see Selective Criteria.

If the screening is successful, an e-mail notification is sent to the specified address.

The service is automatically turned off when there are no active criteria entries.

CONFIGURATION

The user defines the e-mail address where notifications are to be sent and the criteria used to filter incoming calls. The user can selectively activate or deactivate criteria entries.

CALL RECORDING

The Call Recording service allows a user to record their calls.

DESCRIPTION

This service creates connections to a recording platform for the recording of users' calls. Both the originating and the terminating calls can be recorded. If the user is recording a video call, then the video portion of the call is also recorded.

It is possible that more than one user on a call has the Call Recording service activated. In such a case, a separate call recording is made for each user who is recording the call.

The service can record all calls, selectively record calls (or portions of calls) triggered by a user's input, or record no calls that the user makes or receives. The following service modes control the recording behavior:

- If the user's recording mode is *Always*, then the Call Recording service automatically records all users' calls and saves them to the recording platform. Recording starts when the call is answered. If an answer confirmation is required for a call, the recording starts after the call is accepted.

- If the user's recording mode is *Always with Pause/Resume support,* then the service automatically records all the user's calls. However, the user can pause the recording, for example, by dialing *48 (default) and resume a paused recording, for example, by dialing *49 (default). The parts of the call when the recording is paused are not kept.
- If the user's recording mode is *On Demand*, then the Call Recording service records the calls, but only the recordings of calls that the user triggers, for example, by dialing *44 (default) are kept. Notification that the user wants to keep the recording must be sent to Clearspan prior to the call's ending. Otherwise, the recording is not kept. Once the user has indicated that the recording should be kept, the user can pause and resume the recording during the call, to keep only selected parts of it.
- If the user's recording mode is *On Demand with User Initiated Start*, then call recording does not start until the user starts recording the call. Once the call is being recorded, the user can pause/resume and stop the recording. Only the parts of the call selected by the user are recorded. When the recording is stopped, for example, by dialing *45 (default), the connection to the recording platform is disconnected. If the user starts to record the call after stopping a recording, a new recording is generated.
- If the user's call recording mode is other than *Never* and the recording of voice messaging in enabled, then the voice messages left for the user are recorded, including the caller's interactions with the Voice Messaging system. The system behaves as if the mode was *Always* since the user has no way of interacting with the system when the call is being recorded.
- If the user's recording mode is *Never*, then no calls are recorded for the user. Although the service is assigned to the user, nothing is recorded, even if the user attempts to trigger the recording.

The user recording a call can decide to notify call participants that the call is being recorded. If call recording notification is enabled, then callers hear an announcement or a warning tone notifying them that the call is being recorded, or that the recording has been paused or resumed.

The Call Recording service can be assigned to a Clearspan user and to a virtual user, such as call center, route point, or Auto Attendant. The only way to activate the service for virtual users is to set the mode to "Always".

Call Recordings

The call recording starts after the called party answers the call or after the recording party presses the record button. Any announcements or early media played prior to the call being answered are not recorded.

Recording Three-Way and N-Way Conferences

When a user initiates a conference call, all calls participating in the conference (and being recorded) generate their own recordings.

If a call is transferred to a conference, then the recording changes from the conversation between the conference initiator and the party, to the recording of the conference call.

If the conference initiator requests call recording after the conference has started, the recording is started if the conference call is the most recent locally held call. In this case, all of the participating calls are selected and recording is started on each of the calls.

Many devices are capable of setting up three-way conference calls without the need for a network conference bridge. If the user's device bridges the calls together, they appear as two separate call recordings. There is no indication that the two calls are part of a conference call. If Clearspan provides the conference bridge, all of the participants of the conference are listed in the XML extension data of the recordings.

If the user participating in the conference is not the controller, then the recording follows the same rules as those a two-way call.

Recording Parked Calls

If the party being parked records the call, there is a single recording as a result. The recording includes the time the user was parked, as well as any silence or music during the time the user was parked depending on the configuration of the Call Park service.

Recording Voice Mail Calls

The user has the option to record voice mail messages, and the caller's interactions with the Voice Messaging system while callers are leaving messages in the user's mailbox.

Recording Video Calls

This service provides the capability to record the video portion of calls in addition to audio. It interfaces with the recording platform to support audio and video recordings in both single and dual modes. This includes support for the recording of three-way and n-way video conferences in both single and dual modes.

CONFIGURATION

The system administrator provisions the external recording platforms that can be used to record calls and assigns a default platform to service providers/enterprises authorized to use the service. The system administrator also specifies whether the audio is sent to the platform in a single stream (single) or in two separate streams (dual). In the event of a dual stream, the audio from the user who is recording the call is sent in one stream and the audio to the user is sent in the other stream.

The group administrator selects the platform to use for their group.

The user selects the service mode and specifies whether they want calls to their voice mail to be recorded and whether callers should be notified of the call being recorded. Setting the mode to “Never” is equivalent to disabling the service.

CALL RETURN

The Call Return service enables a user to return the call from the last party that called. The system stores the number of the last party that called, and when the user dials a recall feature access code, the system attempts to connect the user to that party.

DESCRIPTION

This service allows a user to call the last party that called by dialing *69 (default) on the user’s device.

The number to call back must be available to Clearspan. If the number is available, the last calling party is called as if the user dialed this number directly. If the number is not available, the user is played an error announcement.

A call originated with Call Return is subject to all users’ services and restrictions.

A system parameter determines whether returning a call to a restricted number is allowed. When it is not allowed and a user tries to use Call Return on a call with the caller ID restricted, the user is played an error announcement.

Depending on system setup, either both answered and unanswered calls can be called back or only unanswered calls can be called back.

The user can delete the last calling number by dialing #92# (default). If only unanswered calls can be called back, the last unanswered incoming number is deleted. If both answered and unanswered calls can be called back, the most recent incoming number (answered or unanswered) is deleted.

Upon successful deletion, a confirmation announcement is played. After an incoming number is deleted, Call Return cannot be used until a new incoming call has been received.

The system can also be configured to delete a phone number automatically when the call returned to that number is answered.

Two-level Mode

The service can be configured to work in a two-level mode, which provides announcements to guide the user. When the user dials the Call Return feature access code, the last incoming number is announced and the user is instructed to dial a

FEATURE	BASIC	STANDARD	PREMIUM
Call Recording	X	X	X
Service Integration			
Voice Mail	X	X	X
Integration with Clearspan User Services	X	X	X
Outlook Contact Integration	X	X	X

Features

This section provides an overview of the functionality provided by the Call Center service.

Note: All functionality is provided by the Call Center – Premium, the most complete Clearspan Call Center offering. For information on functionality provided by each Call Center offering, see the previous section.

Call Distribution to Agents – When at least one agent is available to receive calls, incoming calls are handled according to the selected policy, which can be one of the following: circular, linear (regular), simultaneous, uniform, or weighted call distribution. For more information on distribution policies, see Hunt Group.

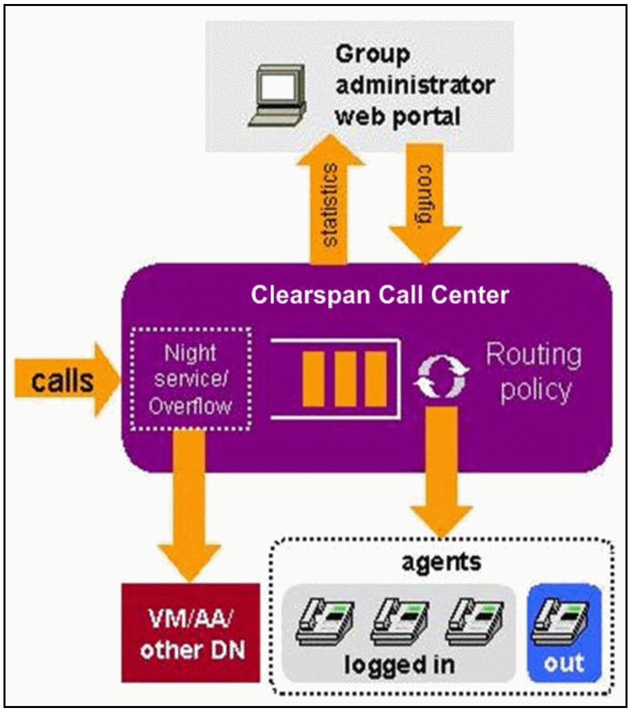

Figure 4 Call Distribution Policies

- **Call Escalation to Supervisor** – Agents can escalate a call to a supervisor by dialing #83 (default). The current call is placed on hold.
- **Emergency Escalation** – Agents can emergency escalate a call to a supervisor by pressing a key. The original call is not placed on hold and the supervisor is immediately conferenced into the call.
- **Multiple Queue Priority Routing** – This feature is used to determine which call to deliver in priority, when an agent becomes available, and calls from several queues served by that agent are waiting to be answered. Bounced calls always have priority over non-bounced calls. If there is more than one candidate bounced call, then they are prioritized based on the original time at which the call was offered to an agent.
- **Uniform Call Distribution Priority Scope** – The Call Center service can use call center or agent scope to determine the next available agent when using the Uniform Call Distribution policy.
- **Call Queuing** – When all call center agents are busy, incoming calls can be queued until they can be presented to an available agent.
- **Distinctive Ringing** – Distinctive Ringing can be provided to an agent when the call is routed from the call center. The Distinctive Ringing policy has precedence over the Priority Alert service assigned and configured to a user who is also an agent. The agent’s Priority Alert service does not modify Distinctive Ringing set by the Call Center service when the call comes from the call center.

  Another distinctive ring can be configured for forced delivery of calls.

- **Call Center Call Information** – A configurable parameter enables Clearspan to send additional call center call information to the agent's SIP phone, including call center name, call waiting time, number of calls in queue, and the longest waiting time.
- **DNIS –** A call center can be associated with up to 63 Dialed Number Identification Service (DNIS) numbers with configurable priority. A priority is attached to an incoming call based on the DNIS number on which it is received, and calls are distributed to the agents based on their priority.
- **Call Statistics Reset** – By default, the call waiting time is preserved when a call is transferred to a new queue. This function allows the resetting of the call waiting time; the call is then treated as a new call and is placed at the bottom of the queue.
- **Overflow** – When a call center cannot accept any more calls or when calls are not handled within a specified amount of time, these overflow calls can be forwarded to a configurable overflow phone number, receive busy treatment, or be provided with a ringing tone.
- **Bounced Calls** – Calls that are delivered to an agent but not answered as specified, are prioritized ahead of other non-bounced calls in the queue.
- **Stranded Calls** – When all agents in the call center group log out, queued calls can be automatically sent to the specified phone number, receive busy treatment, or left in the queue.
- **Forced Forwarding** – Calls to a queue can be temporarily diverted to a configured destination.
- **Holiday and Night Service** – Calls received after hours or on non-business days can be routed to voice message or transferred to a configurable emergency number.
- **Forced Delivery of Calls** – Queues can be configured to answer calls after a specified waiting time automatically and calls can be delivered to agents over the device's speaker and microphone.
- **Call Center Agent Settings** – Administrators can allow call waiting on agents, calls to agents in wrap-up state, and maximum ACD wrap-up time.
- **Basic Announcements** – 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.
- **Comfort-Bypass Messaging** – Calls that are expected to be answered quickly can be provided with an alternate, shorter comfort message instead of the usual comfort/Music-On-Hold treatment.
- **Queue Escape** – Callers who are queued can press a key to be sent directly to the call center voice mailbox instead of waiting for an available agent.
- **Call Treatment Chaining** – Multiple files and URLs can be configured for the call center audio and video treatments.
- **Call Center Support for Agent and Supervisor Clients** – This feature provides support for agent and supervisor client applications. For more information, see *Agent and Supervisor Clients*.
- **Third-Party Call Center clients** – This feature enables the development of third-party Call Center clients that use the call center functionality on the Clearspan

platform. The third-party Call Center clients use the Clearspan Open Client Interface (OCI) to communicate with Clearspan.

- **Geographic Distribution** – The Call Center service allows agents to be geographically distributed. Therefore, agents can attend calls from home, a satellite office, or any other location served by Clearspan in a transparent fashion.
- **Agent ACD State** – Agent state is used to determine the agent's availability to take calls. Agents can set their state through the web portal or through a client.
- **Allow Agents to Join Call Centers** – This setting determines whether agents are allowed to join a call center.
- **Agent Sign-in/Sign-out** – Agents can log in and out from the call center so that calls are only presented to agents who are on duty.
- **OSS Access** – This feature allows call center agents to join call centers to which they are assigned via the OSS if *joining the call center* is enabled.
- **Agent Guard Timer Setting** – The guard timer allows a configurable delay between the time an agent becomes available and the time a call is delivered to the agent.
- **Agent Availability Settings** – Agents can be configured to have their state automatically changed to *unavailable* after a configurable number of bounced calls, upon activation of Do Not Disturb service, or when they are answering or receiving personal calls.
- **Agent Unavailable Codes** – When agents change their ACD status to *unavailable*, they may be required to provide the reason for their unavailability by entering an unavailable code. Administrators can also define default codes to be assigned to agents who become unavailable.
- **Disposition Codes** – Disposition codes are additional attributes that can be applied to calls during the call and during wrap-up. This allows calls to be tagged with promotions, topics, and so on.
- **Agent Hold and Retrieve** – This feature prevents agents from putting a call center call on hold and leaving the caller on hold indefinitely. The agent must take the call off hold or return the call to the queue.
- **Call Center Outgoing Calls** – Agents have the ability to make calls using the call center phone number. The called party sees the calling line ID (CLID) of the call center or DNIS instead of the agent's CLID. Agents can use configurable feature access codes to make calls with either the call center or their own phone number.
- **Agent whisper Message** – A whisper message is played to the agent while the caller receives ringing. The message typically announces from which call center queue the call is coming. This is especially important when using DNIS.
- **Supervisor Support** – Supervisors can be assigned to more than one call center and can select the call centers and agents they want to supervise.
- **Silent Monitoring** – Supervisors can silently monitor an agent's call. The agent may or may not be notified that their call is being monitored. To silently monitor calls, the supervisor must have Directed Call Pickup with Barge-in and Call Center Monitoring services assigned.

- **Call Center Status And Statistics** – Statistics are generated for each call center and each agent in the call centers, and can be viewed by the group administrator via the web portal and/or periodically dispatched to a configurable destination.
- **Call Center Reporting**: Statistics reports can be generated and scheduled in a variety of ways. For more information on the Call Center Reporting service, see *Call Center Reporting*.
- **Call Recording** – Call center calls as well as agent and supervisor calls can be recorded using a third-party recording platform.
- **Service integration** – Any Clearspan personal service can be assigned to a call center phone number to customize the call center. This includes services such as Call Forwarding, Call Notification, Call Screening, and Voice Messaging.
- **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 center voice mailbox.
- **Outlook Contact Integration** – vCards from the agent's Outlook or Exchange contact database appear for incoming calls.

CALL CENTER REPORTING

The Clearspan Call Center Reporting offers three options for queue and agent statistics collection and reporting.

- The basic option offers a limited set of statistics that can be viewed by the administrators on a per-call center basis, by specifying the start and end time of the period for which statistics are requested. Alternatively, a reporting function can be enabled such that e-mail reports are sent on a daily basis. These reports provide statistics with a granularity as small as 15 minutes. The statistics are collected on the Application Server and kept for a minimum of 48 hours. However, the Application Server may also be configured to use the Call Center Reporting Server (CCRS) or centralized database as the source of statistical data.
- The CCRS option is implemented on the Call Center Reporting Server (CCRS). The Call Center Reporting Server receives agent call events and queue events from the Application Server and provides a web service interface to the call center call clients (supervisor/agent) to retrieve the statistics. With this option, the statistics are accumulated by the Call Center Reporting Server and rendered by the Call Center clients via a small number of predefined reports.
- The enhanced option is implemented using a centralized database. It offers a large set of statistics and enhanced reporting capabilities, which include canned and customized templates and scheduled reports. This option is introduced in parallel with the existing options to allow customers to migrate from one solution to another over time.

AGENT AND SUPERVISOR CLIENTS

The Provisioning Server (PS) supports call center supervisor and call center agent clients. Clearspan Agent and Supervisor services are often assigned to agents and supervisors of a call center to enable the use of the Clearspan Call Center client application. To be able to use Call Center client functionality, the call center must be a Standard or Premium type.

Clearspan Agent

Clearspan Agent provides the user with the ability to use the client in agent mode to automatically answer incoming calls. The service can be assigned to any user. However, to use the Call Center functionality, the user also needs a call center license.

A CAP application ID is created for the Clearspan Agent. Clients using the Clearspan Agent as the application ID can log in only with the Clearspan Agent service assigned. These clients have complete ability to use CAP call control messages.

An agent can also use the client to answer incoming calls automatically. This is allowed only if the device supports the remote control talk event package. The Clearspan Call Center client supports this capability.

Clearspan Supervisor

Clearspan Supervisor provides the user with the ability to use the client in supervisor mode to supervise agents and queues and to view statistics. The service can be assigned to any user to enable that user to supervise agents for one or more call centers. A user can be a supervisor for multiple call centers. The number of supervisors with which a call center can be associated is not restricted.

The Clearspan Supervisor service includes the functionality of the Clearspan Agent service. A Clearspan Supervisor is also considered to have Clearspan Agent capabilities. Therefore, to be an agent, a user having Clearspan Supervisor assigned does not need the Clearspan Agent service also assigned to them.

A CAP application ID is created for the Clearspan Supervisor. Clients using Clearspan Supervisor as the application ID can log in only with the Clearspan Supervisor service assigned. These clients have complete ability to use CAP call control messages.

CONFIGURATION

Call centers are created at the group level; however, the service is licensed at the user level. To create a call center of a given type (Basic, Standard, or Premium), the Call Center service of the corresponding type must be authorized for the group. Before agents can be assigned to a call center of a given type, they need to be assigned the corresponding Call Center service license. Supervisors do not need a call center license.

The group administrator configures the group-level service settings and creates call centers as instances of the Call Center service.

Three different types of call centers can be created based on which feature is authorized for the group. Through the *configuration* pages, the administrator can configure call centers and set users who should be part of the call center. Call center configuration includes the following:

- Call center type: Basic, Standard, or Premium
- Basic information, such as call center name, phone number, and queue length
- Call distribution policy (Circular, Regular, Simultaneous, Uniform)
- Call center profile settings such as, the agent's ability to join a call center, whether video announcements are allowed, whether the call waiting time should be reset upon being transferred to a new queue, or whether call waiting on agents is allowed.
- Call routing policies (such as, Overflow, Night service, Bounced Calls, or Stranded Calls) settings
- Call center announcement attributes specifying the source for each type of announcement
- Statistics and reporting settings, such as statistics source, statistics sampling frequency, reports branding, and definition of custom reports
- The agents and supervisors assigned to a call center
- Agents settings and agents unavailable codes
- Services
- Disposition codes that can be assigned to calls
- DNIS numbers

CALL PARK

The Call Park service allows a user to park a call against an extension so that any member of the group or enterprise can retrieve it. While the call is parked, the user can make and receive other calls freely and invoke other features without limitation.

DESCRIPTION

The Call Park service allows users to park a call so that any member of the group or enterprise, if the group is part of an enterprise, can retrieve it with the Call Park Retrieve function.

A call can be parked against any user of the group or enterprise, if the group is part of an enterprise, including the user who parks the call. However, a user can only have one call parked at a time.

Note: A call cannot be parked against a remote party.
There must be only one other active call when the call park is attempted. If there are multiple calls in progress, the system would not be able to determine which call to park.

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 and then the call is parked. (Although entering a full DN is supported, the party against which the call is parked must be in the same enterprise as the party parking the call.) If no number is entered and the user hangs up immediately after dialing the feature access code, the call is parked against the user's line.

An enhancement, Group Call Park, provides a hunting mechanism so that when parking a call, the service hunts for an available user in a configured call park group as a place to park the call instead of only trying the parking user. To use Group Call Park to park a call, the user presses the flash hook during an established call and then dials the Group Call Park feature access code. If no number is entered and the user hangs up immediately after dialing the feature access code, the call is parked against an available user's line.

Note: A call park group is defined by the administrator as a subset of the users in the group, but the users can park calls across groups within the same enterprise.

While parked, the parked party hears the audio on hold configured for that group. (If no audio on hold is configured for the group of the user parking the call, the parked party hears silence.)

To retrieve a parked call, the user dials the Call Park Retrieve feature access code, which results in prompting the users to enter a number where the call to be retrieved is parked. (Although entering a full DN is supported, the party against which a call is parked must be in the same enterprise as the party parking the call.) Upon entering the number, the user is connected to the parked party. If no number is dialed after the feature access code, the user is reconnected to the call parked against the user's line. (To retrieve calls parked against them, users enter the Call Park Retrieve feature access code followed by the #, an inter-digit time-out, or their own extension.)

A configurable recall timer is started when a user parks a call. If the timer expires before the parked call is retrieved, Clearspan determines whether the parking party is idle. If it is idle, the parking party is alerted and the user's phone is rung (if on-hook). The behavior is similar to Hold Recall.

Note: All Call Park recalls are diversion inhibited except if the recall destination is a hunt group.

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.

When a parked call is recalled, the user is provided with an audio and visual indication to be able to distinguish between a reverted call and a new call.

Optionally, a hunt group can be selected as a recall destination and alerted instead of the parking user or if the parking user does not answer the recall.

CONFIGURATION

The group administrator assigns the Call Park service to the entire group at once through the 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.
- Configure a distinctive ring pattern for calls reverted from Call Park.
- Provision an alternative recall destination. This destination must be a hunt group within the same enterprise or service provider as the parking user. An alternative user may be provisioned for the Call Park service and a different alternative user may be provisioned for each call park group.

CALL PICKUP

Call Pickup is a multiuser service that allows selected 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 who 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 is 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.

The Directed Call Pickup user service enhances the Call Pickup service so that its scope encompasses an enterprise as opposed to being restricted to a group.

CONFIGURATION

The group administrator defines call pickup groups through the web portal. A single group can have multiple call pickup groups defined simultaneously; however, a given user can only belong to a single call pickup group.

CALLING PLAN

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

DESCRIPTION

The administrator controls the type of calls made, received, transferred, and forwarded by users in a group. The restrictions are applied by means of sets of call screening templates assigned to groups, departments, or single users. The templates specify various screening methods that should be applied to calls according to the call type or the digits dialed.

The administrator can define different screening templates for incoming, outgoing, redirected, and Call-Me-Now calls. The following subsections describe these capabilities in more detail.

Call Topology

The administrator can define different screening templates for the following:

- **Incoming calls** – The incoming call screening template allows the administrator to define how calls that are received by users should be restricted.
- **Outgoing calls** – The outgoing call screening template allows the administrator to define how calls that are originated 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 the administrator to prevent calls from being forwarded or transferred to external parties, offering Fully Restricted Originating functionality. Note that if either party involved in a call is fully restricted, neither of them is allowed to forward/transfer the call or have their call forwarded/transferred to an external party.
- **Call-Me-Now calls** – This screening template allows the administrator to define how Call-Me-Now calls should be restricted.

The call screening templates apply independently to different legs of the call. For instance, when a call is transferred by the user, both the incoming and transferred call screening templates are applied to the call sequentially.

Call Types

The incoming call screening template 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.
- **Collect calls** – When this option is checked, users are allowed to receive collect calls. When an incoming collect call is blocked, the caller is played an announcement stating that the called party is not authorized to receive collect calls. (Clearspan relies on the *Calling Party Category (CPC)* parameter of the *Generic Transparency Descriptor (GTD)* parameter received in the incoming SIP INVITE message to identify collect calls. When the CPC value is not related to an operator

call or when the *GTD* parameter is not present, it is assumed that the call is not a collect call.)

- **Calls from outside the group** – The “calls from outside group” screening criteria of the Incoming Calling Plan (ICP) provides a distinction between:
  - Allow calls from outside of the group
  - Partial – Allow calls from outside of the group only if transferred by a group use*r*
  - Block calls from outside of the group

For a *user, setting the “calls 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, the caller receives the standard Incoming Calling Plan denial announcement.

Note: In enterprise scenarios, the “Calls from within the group” and “Calls from outside the group” screening criteria refer only to calls from outside the enterprise. Calls from outside the group but within the same enterprise do not trigger the “Calls from within the group” and “Calls from outside the group” screening criteria.

The outgoing, forward/transfer, and Call-Me-Now call screening templates can screen the following:

- **Group** – Calls from within the user’s business group.
- **Local** – Calls within the same geographic region.
- **Toll free** – Free calls to numbers beginning with 1, usually followed by 800, 877, or 888.
- **Toll** – Chargeable calls within the same geographic region.
- **International** – Chargeable calls to other countries.
- **Operator assisted** – Calls made with the chargeable assistance of an operator.
- **Chargeable directory assistance** – Chargeable calls made to Directory Assistance such as 411 or 555-1212.
- **Special services I** – Calls to 700 numbers. These calls may or may not be chargeable.
- **Special services II** – (Reserved for system administrator’s discretion.)
- **Premium services I** – Chargeable calls to 900 numbers.
- **Premium services II** – Chargeable calls to 976 numbers.
- **Casual** – 1010XXX chargeable calls, such as 1010321.
- **URL** – Chargeable calls made to an e-mail address instead of a phone number.
- **Unknown** – Calls to unknown call types.

Note: When the Network Server is configured to return call type information as part of the list of contacts in response to a routing request, Outgoing Calling Plan rules are applied to toll calls. This is to ensure that dispositions (allow, block, collect authorization code, and transfer to number 1, 2, or 3) are properly applied to toll calls when usual Outgoing Calling Plan analysis (based on pattern matching of dialed digits) has let a call go through, but where the Network Server later identifies the call as a toll call.

The call screening template, being forwarded or transferred, can screen calls from outside the group.

The system administrator defines a digit map for the system that defines digit strings that should be mapped to each call type. For instance, this digit map would assign 911 to emergency calls in North America.

In addition to fixed call types, the Calling Plan service allows the administrator to screen calls against configurable digit strings and pinhole digit strings. The 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 can selectively assign them to the group, to selected departments, or to selected users. These strings can be used to complement the outgoing, forward/transfer, and Call-Me-Now call screening templates. Digit strings are used to further restrict calls, whereas pinhole digit strings are used to allow some calls that would have otherwise been denied by the Calling Plan.

Basic and Enhanced Screening Options

The Calling Plan offers basic and enhanced screening options. The enhanced screening options apply only to the outgoing call screening templates. The remaining call screening templates are 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 outgoing 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; otherwise the call is blocked as described above. (The authorization code entered by the user is also captured in the accounting call detail record generated by Clearspan.)

- **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 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 their 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 is captured in the CDR associated with that call.

Interaction between ODP, OCP, and OPDP

The following outgoing calling plans are available on Clearspan:

- The Outgoing Calling Plan (OCP) denies call originations, forwards/transfers, being forwarded/transferred, and/or Call-Me-Now calls based on call types.
- The Outgoing Digit Plan (ODP) denies call originations, forwards/transfers, and/or Call-Me-Now calls based on digit strings.
- The Outgoing Pinhole Digit Plan (OPDP) allows call originations, forwards/transfers, and/or Call-Me-Now calls based on pinhole digit strings.

The Outgoing Pinhole Digit Plan is applied first, the Outgoing Calling Plan is applied second, and the Outgoing Digit Plan is applied after both of these. When a pinhole digit pattern is matched, Clearspan can allow a call that might otherwise be blocked by the OCP or ODP.

Note: A system-level parameter provides the ability to control how the Outgoing Calling Plan and Outgoing Digit Plan restrictions apply to transferred calls. By default, both legs of the call transfer are screened. For example, if User B initiates call transfer from User A to User C, then the OCP screens the User B to User A leg and the User B to User C leg. However, when *Direct Transfer Screening* is enabled, only the second leg of the transfer is screened, namely the User B to User C leg in this example.

The Outgoing Calling Plan and Outgoing Digit Plan require the Outgoing Calling Plan service is to be authorized and assigned to the group. The Outgoing Pinhole Digit Plan requires the Enhanced Outgoing Calling Plan service to be authorized and assigned to the group in addition to the Outgoing Calling Plan.

Interaction with Virtual Users

- Redirecting services triggered by virtual users are not subject to the Outgoing Calling Plan and the Outgoing Digit Plan, except for Meet-Me Conferencing virtual users.

CONFIGURATION

The group administrator configures the Calling Plan service in a hierarchical fashion. The configuration data is the same in the group, department, and user levels.

- If required, the administrator defines digit strings and pinhole digit strings.
- The administrator first defines default calling plans for the entire group. This plan applies to any department or user who does not have their 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 who belong to that department.
- The group administrator can define a specific calling plan for selected users. In this case, the users' calling plans have precedence over the department and group calling plans.

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.

CONFIGURABLE FEATURE ACCESS CODES

Clearspan allows group administrators to select the FACs used to activate, deactivate, and program various Clearspan services.

DESCRIPTION

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

A feature access code is defined as a string of two through five digits and special characters that are associated with a Clearspan service or function, which is dialed 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 through 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 through 9.

emergency call is permitted or denied, as long as an address is configured and the option of sending e-mails is enabled.

The e-mail includes such information as the date and start time of the call, the user's ID and number, the group's ID and number, the dialed digits, the IP address of the calling device (whether the call was in the zone, and whether it was permitted or denied).

CONFIGURATION

The service can be enabled or disabled by the service provider. The service provider also configures the following

- Home zone – A home zone is a list of IP addresses and/or IP address ranges.
- Reject SIP registrations and outgoing calls outside home zone.
- Deny SIP emergency calls from outside home zone.
- Configure emergency e-mail.

EXTENSION DIALING

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

DESCRIPTION

Extension dialing allows a user to dial an abbreviated digit string to call another user in the user's group. By default, the extension is set to the last *n* digits of the user's phone number. (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.) However, the group administrator can change it to any other valid string of digits that is not already in use by another member of the group.

Once assigned, users' extensions can be used for dialing and for intragroup routing applications that require a phone number (for example, Call Forwarding, Simultaneous Ringing, Speed Dial, and so on).

Extensions can be *dialed* from the phone.

Callers to the group Auto Attendant can use the dial by extension option to reach any user of the group through their extension.

Extensions can be assigned to users, external users, and virtual users. Users without a phone number can have just an extension. For more information about assigning extensions to external users, see Virtual On-Net Enterprise Extensions.

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

CONFIGURATION

This service does not have to be authorized or assigned to groups.

The system administrator defines the maximum and minimum extension length allowed in the system, which can be from two through 20 digits.

The group administrator can modify these settings and can set the default extension length for their group, within the limits set by the system administrator.

- If the group administrator sets the minimum and the maximum lengths to different numbers, users in the group can have extensions of different lengths.
- Each user and virtual user's extension is populated by default with the last *n* digits of the user's phone number, where n is equal to the default extension length set by the group administrator.
- For users without phone numbers and for other cases where the default extension is not appropriate, the group administrator can set the extension through each user's and virtual user's configuration page.
- The group Auto Attendant and the group (or enterprise) directory are provisioned with the extensions of the users within their scope automatically.

FIND-ME/FOLLOW-ME

The Find-me/Follow-me (FMFM) service is used to dispatch incoming calls across multiple family devices.

For a given Find-me/Follow-me group, it is possible to create an Alerting policy that combines Sequential and Simultaneous Ringing. The policy can be applied to the members of the family (that is, Clearspan group members) and addresses (that is, phone numbers, extensions, and SIP URIs).

This feature also adds the ability for members of the Find-me/Follow-me group to perform a Call Push to send a call back to the Find-me/Follow-me group so that another member of the group can answer the call.

DESCRIPTION

The Find-me/Follow-me service allows a group administrator to define a Find-me/Follow-me group and a set of alerting groups through which to route inbound calls received by the Find-me/Follow-me group. The Find-me/Follow-me group is assigned an address. When a call is placed to this address, the call is routed through the alerting groups defined for this Find-me/Follow-me group. Each alerting group contains a list of users and external devices to ring simultaneously. Each alerting group can also be assigned selective criteria to determine whether the inbound call should be distributed to this alerting group. The service sequentially advances though the list of alerting groups until the call is answered or all groups have been tried.

For the Find-me/Follow-me group calls, the *voice mail no answer* timer is ignored to give all of the alerting groups a chance to process the call.

When the call is presented to the members of the alerting groups, the calling line identification (CLID) information presented is that of the originator of the call, and not the Find-me/Follow-me group identity. This allows the called party to identify who is calling. However, the CLID information is still subject to the usual service controls for this information, such as Calling Line ID Blocking, Calling Line ID Delivery, Calling Name and Number Delivery, and so on.

In addition, the Diversion Inhibitor and answer confirmation can be enabled or disabled for each alerting group.

Call Push

The Find-me/Follow-me service allows a user who is a member of a Find-me/Follow-me group to push calls received from the group, back to the group. Pushing the call back to the group causes the call to be processed by the Find-me/Follow-me group again to re-alert its members. This functionality is limited to calls received from the Find-me/ Follow-me group by Clearspan users who are part of that Find-me/Follow-me group. Any Clearspan user who is a member of at least one Find-me/Follow-me group is automatically provided with the Call Push functionality.

The Call Push is initiated by a user placing the call on hold and then initiating a call to Call Push feature access code (FAC). This causes the call to be redirected to the Find-me/ Follow-me group and to be routed through all of the alerting groups again. The user who pushed the call is not re-alerted after the call is pushed. If there are multiple redirections, the users pushing the call can be re-alerted depending on the redirection sequence.

CONFIGURATION

Find-me/Follow-me groups are virtual users created at the group level. The group must have the Find-me/Follow-me service assigned and activated to be able to create Find-me/ Follow-me groups.

The administrator configures Find-me/Follow-me through the group web portal.

The configuration includes the following:

- Basic information, such as name, calling line identity, department, and time zone
- Phone number and/or extension
- Alerting groups

- Services assigned and their settings

For each alerting group, the group administrator configures a set of Clearspan users and external devices to ring simultaneously, defined a set of selective criteria to determine whether a call applies to the group, and enables or disables Diversion Inhibitor and/or answer confirmation.

GROUP CALLING LINE IDENTITY

This service allows a calling line identity (name and number) to be defined for an entire group and used as calling line identity by the group's users.

DESCRIPTION

This service allows the group administrator to define a default group calling line ID.

The default group number applies to all external calls made by non-DID users (also known as extension-only users). (The default group CLID is never used for intragroup calls.) It also provides a default billing number for non-DID users, thus allowing them to make external calls.

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 of the group CLID for that user is blocked.

The DID used as a default group CLID can still be assigned to a user in the group. For instance, it can be assigned to a group Auto Attendant to allow external parties to use it to reach non-DID users (phantom users).

In addition, for users assigned to a department with a CLD number defined, the department phone number can be used instead of the group phone number as the calling line identity.

CONFIGURATION

This service does not have to be authorized or assigned to groups.

The group administrator defines the group CLID name and/or number and configures the Calling Line ID Call Processing policy for the group (and/or individual users) to specify whether group or user CLID should be used as calling line identity.

Note: The DID selected as the group calling line ID number is still available to be assigned to a real or virtual user of the group.

GROUP PAGING

Group Paging is a virtual user service that allows for unidirectional paging to a group of users by dialing a paging group phone number or extension.

DESCRIPTION

This service allows a user to unidirectionally page a predefined group of users by dialing a phone number or extension assigned to that group.

A paging group is an instance of the Group Paging service and is assigned a list of targets, a list of originators, and a phone number/extension.

Only the originators are allowed to use the phone number assigned to the paging group. When an originator dials the paging group phone number, all the targets are paged and connected into an “n-way” conference with the originator.

Paging groups are included in the phone directory based on the privacy settings.

The maximum number of targets in any paging group is controlled at the service provider, enterprise, and group levels. The maximum number of targets that can be configured in any paging group is 1,000.

Nested Paging Groups

A paging group can be defined as a target in another paging group, thus creating a nested paging group. The nesting is limited to a single level.

The total numbers of users in a nested paging group must comply with the configured maximum.

A paging group cannot be deleted from the system if it is nested inside another paging group.

Targets

Any user or paging group within a group or enterprise can be a target in a paging group. Virtual On-Net users are not possible targets.

In addition, the user can be a target in multiple paging groups.

Group Paging also applies to the target user’s alternate locations. A configuration option is provided at the user level for services such as Shared Call Appearance and Clearspan Anywhere to enable alternate locations for paging.

Diversions toward the targets are inhibited for a page. Therefore, Forking services and Call Forwarding services assigned to the target are ignored.

A page is initiated to a target if the target is able to receive the page, that is, if the target is “idle” or has Call Waiting enabled. If the target is busy or has the Do Not Disturb service enabled, they are not paged.

Originators

Any user within a group or enterprise can be an originator in a paging group. Virtual On-Net users are not possible originators.

In addition, the user can be assigned as an originator in multiple paging groups.

If the originator is also included in the paging group as a target, the originator is excluded from the list of targets to be paged when they make a call to the paging group.

Services

Being a virtual user, a paging group can be assigned services. Only the following services are assignable to a paging group:

- Call Notify
- Custom Ringback
- Privacy
- Selective Call Acceptance
- Selective Call Rejection

CONFIGURATION

Paging groups are virtual users created at the group level. The group must have the Group Paging service assigned and activated to be able to create paging groups. Each paging group consumes one Group Paging service license.

The administrator configures paging groups through the group web portal.

The configuration includes the following:

- Basic information, such as name, calling line identity, department, and time zone
- Phone number and/or extension
- Targets and originators
- Services assignment and settings

At the service provider, enterprise, and group level, limits are set for the maximum number of targets allowed in a paging group.

At the system level, the administrator needs to set the *supportAnswerAfter* setting to "true" at the following Application Server CLI level: *AS_CLI/Interface/SIP*. This ensures that the INVITE messages sent by the paging group to the target devices contain the *Call-Info* header with answer-after set to "0", which tells the target devices to auto-answer the page.

HUNT GROUP

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.

DESCRIPTION

The Hunt Group service allows for the processing of a high volume of calls to a single phone number by distributing the incoming calls to multiple users according to a selected 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 on an ordered list. The hunting process essentially determines how to process that list to find an idle user where the call can be terminated.

Clearspan supports the following Hunting policies:

- **Regular (linear)** – The incoming calls to the group start hunting on the first user on 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 on 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 who 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.

A hunt group can redirect calls to the next agent if not answered in a specific number of rings by the previous agent, or forward calls to a specified number if not answered within a specific number of seconds. However, if all agents are found to be unreachable before the forwarding timer expires, then the call is provided with busy treatment.

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.

This feature provides the capability to enable Call Waiting for hunt group agents.

User Services

As hunt groups are virtual users, they can be assigned some users services, for example, Anonymous Call Rejection, Directory Number Hunting (see Directory Number Hunting), certain Call Forwarding services, Do Not Disturb, Voice Messaging, and others.

In addition, the Hunt Group service allows Call Waiting to be enabled for hunt group agents.

Interactions Between User Services and Hunt Group

Users who are members of a hunt group can have their own phone number where they receive calls and their own services 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 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. This includes forwarding services such as Simultaneous Ringing.
- **Voice Mail** – Incoming calls to the hunt group directory number (DN) are never forwarded by the Voice Mail service assigned to a member of the hunt group.
- **Call Transfer** – A member of a hunt group can transfer/blind transfer the call via the phone.
- **Remote Office** – The Remote Office service assigned to a member of a hunt group is honored as usual when a hunt group termination occurs to this user.
- **Clearspan Anywhere** – The Clearspan Anywhere locations configured for a user who is a member of a hunt group are alerted as usual when a hunt group termination occurs to this user.

Directory Number Hunting

Directory Number Hunting is a service extension that allows a caller to reach a hunt group (or a call center) by calling the number of one of the hunt group users. When Directory Number Hunting is enabled and a hunt group user receives a call, the Hunt Group service directs the call to the called user first. If this user is unavailable, the service then applies the Distribution policy that has been configured for the hunt group. For example, for the Regular (Sequential) Hunting policy, the called user is skipped and for the Simultaneous Hunting policy, the hunt group alerts all users simultaneously (as if the caller had called the pilot number).

Note: Directory Number Hunting can only be assigned to hunt groups or call centers that are virtual users. The service does not apply to regular users and cannot be added to a service pack.

Hunt Group Busy

The Enable Group Busy policy can be used to make the hunt group busy. If allowed, the members of the hunt group can turn this policy on or off. Otherwise, only the administrator can make the hunt group busy.

CONFIGURATION

A group administrator creates hunt groups as instances of the Hunt Group service. 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 can be configured for a hunt group:

- Basic information, such hunt group ID, name, phone number, department, calling line identity, and other attributes typically configured for a user. Depending on the calling line ID settings, the hunt group name may be prefixed to the caller ID delivered to the hunt group member when a call terminated through a hunt group. As a result, if a call from Bob Smith is presented to a user through the “Support” hunt group, the CLID appears as “Support – Bob Smith” on the user’s phone.
- The Call Distribution policy used to distribute calls to members.
- No answer, not reachable settings that specify how calls should be processed when the hunt group is not reachable or there is no answer.
- Call Waiting
- The members of the hunt group selected among users of the group. The members are provisioned on an ordered list.
- Calling plans

In addition, certain user services can be assigned and configured for the hunt group.

If the Directory Number Hunting service is assigned to a hunt group, the administrator selects the users for Directory Number Hunting from the full list of the hunt group’s users. A user can be a member of only one Directory Number Hunting group.

The Enable Group Busy policy can be turned on or off.

INSTANT GROUP CALL

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

DESCRIPTION

This service allows an administrator to define a group of users to be alerted simultaneously when a call is made to the group. 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).

CONFIGURATION

A group administrator creates an instance of the service and sets its attributes.

The following can be configured for a service instance of Instant Group Call:

- Basic information, such ID, name, phone number, calling line identity, department, and other attributes typically configured for a user.
- Maximum time to wait for the members to answer the call.
- A list of up to 20 members who are alerted when the virtual user phone number or extension is dialed. A member is defined by the address that is used to reach the member. The address can be a SIP-URI, a phone number, location code and extension, and extension or E.164 number. With the exception of the SIP-URI, the address can be prefixed with an FAC, providing that the corresponding service is assigned to the instant call group.
- Calling Plans

In addition, certain user services can be assigned and configured for a service instance.

INTERCEPT GROUP

The Intercept Group service allows the system to intercept calls routed to a line that has been decommissioned, providing an informative announcement and alternate routing options (for example, “This number is no longer in service. To talk to an operator, press 1”).

The Intercept Group service intercepts calls directed to users within the specified group. The related Intercept User service intercepts calls directed to individual users.

DESCRIPTION

This service allows the administrator to gracefully take a group out of service while providing callers with informative announcements and alternative routing options. Depending on service configuration, none (partial intercept), some, or all incoming calls are intercepted.

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.

The system administrator can define a list of phone numbers allowed to place calls to users in an intercepted group (such as the service provider's customer care number).

In the partial intercept scenario, users can receive calls; however, their ability to make calls is restricted or denied. Partially intercepted users can be provided with a blocking announcement that differs from that for fully intercepted users.

Terminating services are applied to users in partially intercepted groups. This includes services that redirect the call to another number such as but not limited to, the Call Forwarding services (for example, Call Forwarding Always or Call Forwarding Busy), Simultaneous Ringing, and Clearspan Anywhere. The redirected outgoing call initiated by these redirecting services still takes place even though Intercept User is configured to block outgoing calls.

Outgoing calls can be intercepted or rerouted to a configurable customer care number after the intercept announcement is played. Alternatively, the user may be allowed to make local calls.

Emergency and repair calls are permitted, although the system administrator may restrict these call types as well.

CONFIGURATION

The service provider/enterprise administrator configures the service for the group. The configuration includes the following:

- Selecting inbound calls to intercept – All calls can be blocked, all calls can be allowed (partial intercept), or calls from specific system-defined numbers can be allowed
- Selecting outbound calls to intercept – All calls can be blocked, calls can be routed to a configurable phone number, or local calls can be allowed
- Specifying whether mobile calls (inbound and/or outbound) should exempt from being intercepted

- Specifying treatment to apply to allowed incoming calls:
  - Parallel ringing to the intercepted user's alternate network locations can be enabled or disabled
- Specifying treatment to apply to intercepted incoming calls:
  - The caller can be played an out-of-service announcement.
  - The call can be routed to the intercepted user's voice mail.
  - The caller can be played an out-of-service announcement complemented with the playback of the user's new phone number.
  - After hearing the new number, the caller can press a digit to be immediately transferred.
- Specifying treatment to apply to intercepted outgoing calls:
  - The user can be played an out-of-service announcement.
  - The user can be routed to a configurable number after hearing the intercept announcement.
  - In partial intercept, the user may be configured to hear a different announcement when they try to place a call.
- Optionally, replacing default announcements with custom announcements.

The system administrator may choose to restrict intercepted users from making emergency and repair calls. This setting can be configured through the web or CLI, and it also applies to the Intercept User service.

LOUDSPEAKER PAGING

The Loudspeaker Paging service allows users to access an intercom paging system by dialing an extension within the group.

DESCRIPTION

This service allows users in a group to call a number or extension to voice a message over a loudspeaker system.

To voice a message, users simply dial the number or extension associated with the loudspeaker system, and they are presented with a tone indicating that they are "on the air". The connection to the loudspeaker paging system remains until the calling party releases it.

The loudspeaker paging system is provided by a third party.

CONFIGURATION

The Loudspeaker Paging system is configured similar to a regular Clearspan user; however, the following services are suggested:

- By assigning only an extension to the loudspeaker paging system, external parties are prevented from accessing it.
- The Incoming Calling Plan service can also be used to prevent the loudspeaker paging system from receiving external calls.
- The Selective Call Acceptance service can also be assigned to the loudspeaker paging system to prevent the system from being used outside of normal business hours.

MEET-ME CONFERENCING

Clearspan Meet-Me Conferencing provides superior functionality over the Legacy Audio Conferencing application offered by Clearspan. In addition to the many features provided by Clearspan Meet-Me Conferencing, it also offers high definition (HD) audio, which was not available in the legacy Audio Conferencing application. For those customers with the Legacy Audio Conferencing application, a migration tool is provided to convert the legacy “Instant Conference” bridges to “Meet-Me” conference bridges.

DESCRIPTION

The Meet-Me Conferencing service provides the following capabilities:

- Up to 294-way audio conferencing
- Scheduled and reservationless conferences
- Custom greeting
- Recording of conferences
- Muting participants and inviting new participants (by moderator)
- Automatic Lecture Mode
- Web interface to moderate the conference
- Outlook add-in to create conferences from Outlook and including details in meeting invitations
- High definition (HD) audio
- Enhanced security
- Migration tool to convert customers who are using the previous Conferencing solution

A group administrator creates a conference bridge and designates Clearspan subscribers who can host conferences on that bridge. Hosts can create scheduled and reservationless conferences. When a conference is created, there is a host PIN generated along with the conference ID. Any participant who joins the conference using the host PIN has special moderator privileges for that instance of the conference. In addition, the conference host can require that participants enter a security PIN when joining the conference.

Within a conference, moderators can invoke functions such as recording, locking a conference, inviting a new participant by calling the participant from the conference, and so on. There can be multiple moderators for an instance of a conference.

When more than 147 participants join the conference, Automatic Lecture Mode starts, muting all participants except the moderator. Automatic Lecture Mode cannot be turned off, but the moderator can individually unmute up to 100 participants.

The following functions are available to moderators:

- Lecture Mode – The Lecture Mode mutes all participants except for the moderator who turned on the lecture mode. Any participant joining the conference is automatically muted.
- Record Conference – A moderator can record a conference for up to 12 hours. After the recording has been started, it can be paused, restarted, and stopped. When the recording is stopped, the recorded audio is uploaded to the Profile Server and made accessible through the web portal of the Clearspan subscriber who created the conference.
- Invite New Participant using Outcalling – A moderator can originate a call from a conference to a new participant by entering the participant's phone number. Services such as Outgoing Call Restrictions and Conference Bridge-Originating services apply to the originated call. When the called user answers the call, a message is played, inviting the user to join the conference.
- Lock Conference – When a conference is locked, new participants cannot join the conference; however, a moderator can still invite new participants using Outcalling. The conference must be unlocked before new participants can join the conference again. When the last moderator of a conference leaves the conference, the conference is automatically unlocked. This way, a new moderator can join the conference and take control of the conference.
- Transfer to Operator – Upon invoking this option, the moderator is disconnected from the conference and is transferred to the operator.
- Mute – The moderator can mute their line. All other participants can still talk.
- Participant Count and Roll Call – The moderator can obtain a count of the number of participants. If the option to record names when joining the conference is enabled, the names of the participants are played until the moderator presses a key. The Roll Call functionality is disabled when there are over 20 participants in a conference. The list of participants is still available from the *Moderator's Web Control* page.
- Moderator Client and DTMF Menu – These include control functions available to moderators of a conference through the Moderator Client application and a DTMF menu.

A DTMF menu is also available to conference participants, providing such functions as mute, login as moderator, participant count, and roll call.

Meet-Me Conferencing Outlook Add-in is an add-in that enables a user to quickly schedule a Clearspan conference from within Outlook, while they are creating a meeting or an appointment. The invitation for the meeting then includes the conference access information, including URLs to automatically connect to the meeting (if configured).

For more information, see the Clearspan Meet-Me Conferencing Guide.

CONFIGURATION

The system administrator configures general Meet-Me Conferencing setting such as the length of the conference ID, moderator PIN, security PIN; maximum conference duration and expiry date; the length of time the expired conferences are stored; and the URL where the conference recordings are stored. The system administrator also assigns the Meet-Me conference ports to service providers and enterprises, and the service provider and enterprise administrators assign ports to groups.

Group administrators create conference bridges and assign users who can host conferences on those bridges.

Conference hosts create and manage conferences and conference recordings. In addition, the user who created the conference can delegate some of the conference responsibilities to other hosts on the same bridge. Conference hosts and delegates can create custom greetings for the conference.

MUSIC/VIDEO ON HOLD

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

DESCRIPTION

Music On Hold is a group service that allows the group administrator to set up a media audio source that can be broadcasted to held parties in various scenarios. The add-on Music On Hold – Video service allows the administrator to set up a media video source.

The service can be individually enabled or disabled for the Call Hold, Call Park, and Busy Camp On services. In addition, an alternate source file can be specified for internal calls.

When no media file is specified or if Music/Video On Hold is turned off for a service, the remote party hears silence.

Music/Video On Hold settings can be configured on a per-department basis. Once a department is allowed to use their own Music/Video On Hold, the department administrator can configure the service for their department.

Departments without their own audio/video source make use of the group-defined source.

Call centers have their own announcements (including Music/Video On Hold) that are independent of the group Music/Video On Hold service.

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

- **Media source component** – These media files are played back to held parties of the applicable services. The media file that is 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/Video On Hold so that parties held through these services are played back the configured media source.

Media Source

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

Alternatively, the group administrator can configure the Music On Hold service to make use of an external audio source. In this case, the administrator can select the device that provides the audio from among the list of authorized devices for the group.

The audio source is controlled by the enterprise and is typically located on the enterprise's premises. Clearspan 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: To provide Music On Hold, this method only supports audio media.

Users can select a different media source to play to their callers.

CONFIGURATION

The group administrator selects the audio and/or video source for their group and selectively activates or deactivates the service for Call Hold, Call Park, and Busy Camp On services. Optionally, they specify an alternate media source for internal calls.

The group administrator can assign Music/Video On Hold to selected departments and then department administrators can configure the service for their departments.

Users can override the group/department settings and specify their own audio and/or video source.

OUTSIDE ACCESS CODE

Clearspan provides the ability to support PBX-dialing transparency or private dialing plans. Using an access code (for example, 9+), Clearspan can support both a private and public dialing plan simultaneously.

DESCRIPTION

When a user is configured to use an access code (at the system or group level), Clearspan creates an implicit digit map, which contains a digit map for the extensions in the group, feature access codes, and the outside access code.

Clearspan sends this implicit map to the user devices and Media Servers to collect digits, as appropriate.

If the outside access code is reported, Clearspan sends an additional digit map that contains the public dialing plan map. This new digit map is used to collect a public number.

CONFIGURATION

The access code is configurable on a system-wide and per-group basis. The following attributes are configurable:

- System-wide configurable digit map
- Per-group configurable digit map
- System-wide access code
- Per-group access code

Due to the intrinsic nature of protocols, only the Media Gateway Control Protocol (MGCP) device digit maps can be dynamically updated to use outside access codes. SIP access devices support outside access codes; however, their digit map is configured as part of a separate process.

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. It allows this group administrator to 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 specified address as an ASCII comma-separated value (CSV) attachment. The resources reported include:

- Phone numbers
- Devices
- Users and departments

- Services

CONFIGURATION

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

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.

DESCRIPTION

The Series Completion service provides a special hunting capability that is well suited to support a key telephone system (KTS).

The following figure shows a typical Clearspan configuration supporting a three-line key telephone system.

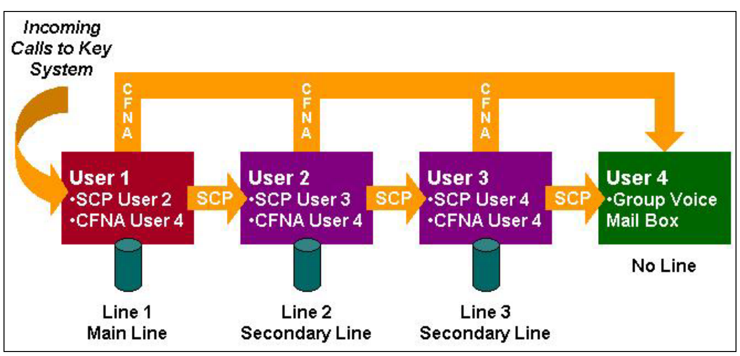

Figure 5 Clearspan Configuration for Three-Line KTS

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. As a result, a series completion group can be viewed as a call forward busy chain, among selected members of a group.

In a scenario in which a customer uses a key telephone system as customer premises equipment, the key telephone system lines can be placed in a series completion arrangement to allow incoming calls to any line (or key) to hunt for the next idle line.

When using series completion to support a key telephone system, other complementary services assigned to users in the series completion group are used by the service, since Series Completion uses user services, unlike Hunt Groups.

CONFIGURATION

The group administrator creates an ordered list of users making up the series completion group.

VIRTUAL ON-NET ENTERPRISE EXTENSIONS

The Virtual On-Net Enterprise Extensions service allows users to call certain external numbers by dialing an extension.

The scope of this service is enterprise for enterprise users and group for service provider users.

DESCRIPTION

The service integrates the virtual private network (VPN) destinations with the Clearspan enterprise framework by explicitly defining external destinations in the enterprise or group directory and assigning extensions to them. Users with the Virtual On-Net Enterprise Extensions service enabled can place calls to these off-net destinations by dialing an extension as they would to members of their group or enterprise, and have special billing applied to these calls.

For service provider users, an external number is a number outside the user's group, whereas for enterprise users, an external number is a number outside the user's enterprise.

The external destinations are associated with first and last names. When presenting users with the calling party or connected party identities of these external destinations, the Virtual On-Net Enterprise Extensions service overrides the public presentation of these destinations with their Virtual On-Net representations provisioned for this service.

The off-net destinations are tagged with configurable Virtual On-Net types for the purpose of differentiated billing.

Virtual On-Net users are automatically included in enterprise and group directories and are visible to users who have been assigned the Virtual On-Net Enterprise Extensions service.

CONFIGURATION

The system administrator defines Virtual On-Net types at the system level. A Virtual On-Net type consists of a mandatory free-form code (CDR value) limited to letters and digits as well as a mandatory name that serves as the label. Predefining Virtual On-Net types allows for the controlling of types in use on the system and ensures consistency across the entire system.

Group administrators provision Virtual On-Net users for the group using the web portal. Virtual On-Net users can be provisioned individually or by specifying ranges of DNs and extensions to create multiple entries. Once the Virtual On-Net user is created, the administrator can change most of the Virtual On-Net user's settings.

The following information must be specified for a Virtual On-Net user:

- Display first and last name
- Calling line ID first and last name
- Public phone number in E.164 format
- Extension (unique within the group or enterprise where the Virtual On-Net user is being defined)
- Virtual On-Net type, selected from the predefined list configured by the system administrator

Virtual On-Net users can be added to custom contact directories.

The service has no configuration at the user level.

MESSAGING

FAX MESSAGING

The Fax Messaging service allows the user to retrieve fax messages from their voice mailboxes and/or e-mail accounts. The service is an add-on to the Voice Messaging service.

DESCRIPTION

Fax messages in the voice mailbox are treated similarly to voice and video messages; users can listen to the headers or envelope of a message (the calling number, the date and time it was recorded, and the number of pages), delete the message, or forward the message to another mailbox. The user may also print the fax message by forwarding it to a phone number terminating at a fax device.

Fax messages sent by e-mail are converted to the TIFF image format and are attached to the e-mail, similar to voice and video messages.

Notifications of new fax messages, such as new audio and video messages, are sent to the message waiting indicator of the user's phone if the phone supports the message waiting indicator and to the user's e-mail account if the user has e-mail notification enabled.

CONFIGURATION

The administrator sets a fax number for the user. The number is taken from the pool of directory numbers available to the user's group. Once configured as a fax number, this number is unavailable for assignment to any other service or profile.

The administrator can also configure up to three fax aliases. Any fax call to one of these aliases is considered by the system as an incoming fax call for the user.

The user cannot modify the fax number or aliases; however, the user can view them through the web portal.

The user can enable or disable the service through the web portal.

DIRECT VOICE MAIL TRANSFER

The Direct Voice Mail Transfer service allows the user to transfer a held remote party (or parties) to the voice mailbox of any user in the user's group, at any time during a call. The target mailbox may be the user's own mailbox.

DESCRIPTION

This capability can be used when the remote party is on hold (that is, consultation hold). From a new consultative call, the user dials the Direct Voice Mail Transfer feature access code to initiate the transfer. A first-time user is guided by announcements that explain how to transfer the held party to the user's or anyone else's mailbox. Power users have the option of dialing through and performing the transfer without waiting for the prompts.

CONFIGURATION

This service has no configuration.

SMDI MESSAGE DESK

The SMDI Message Desk service is a user service that is assigned to a hunt group to support a Legacy Voice Mail system (VMS) accessed over an analog interface.

For calls terminating on the hunt group, the system sends redirecting information to the Legacy Voice Mail system using Simplified Message Desk Interface (SMDI) over a serial interface. This information (calling number, called number, redirection information) can be used by the voice mail system to redirect the calling party to the user's mailbox and provide the correct greeting.

DESCRIPTION

This service allows Clearspan to interface with an external VMS accessed over an analog interface. The analog lines of the legacy VMS are connected to an access gateway hosted on Clearspan. Each analog line is mapped to a user account on Clearspan and the Voice Mail server is the phone number of a hunt group in Clearspan. SMDI Message Desk provides the Voice Mail server with redirection information for incoming calls.

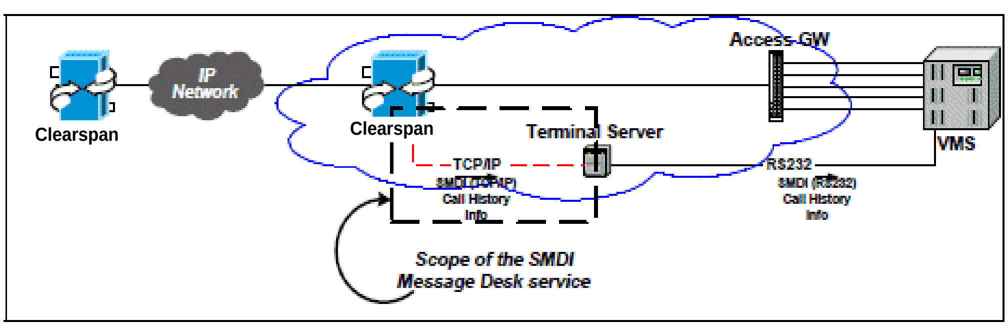

Figure 6 SMDI Message Desk Service

The hunt group delivers the SMDI call history information that ultimately tells the VMS which analog line has been identified as available by the hunt group and selected to deliver the call to the VMS. The SMDI information is delivered over TCP/IP to a terminal server, which in turn delivers it to the legacy VMS over an RS-232 link.

The hunt group sends an SMDI message to a terminal server when any call (from the hunt group to a user account) is alerted.

CONFIGURATION

Note: SMDI Message Desk can only be assigned to a hunt group, which is a virtual user. The service does not apply to regular users and cannot be added to a service pack.

The group administrator performs the following configuration steps:

- Configures an access gateway for the group
- Defines a user account for each line of the VMS

  Each of these accounts is assigned a line on the access gateway.

- Defines and configures a hunt group used to select lines on the VMS:
  - Adds the user accounts representing the lines of the VMS to the list of hunt group members
  - Assigns a phone number to the hunt group
  - Sets the Hunting policy; this application typically uses the “Regular” Hunting policy, however, all Hunting policies, except for “Simultaneous”, are supported.
- Assigns the SMDI Message Desk service to the hunt group and configures it as follows:
  - Enables the service
  - Assigns a three-digit message desk number

  This number is included as a field in the SMDI Call History Messages, and is used by the VMS to identify the hunt group uniquely (within the scope of the service provider).

  - Configures the terminal servers to which SMDI messages are sent

  A terminal server is defined by a name, IP address, and port.

- Configures the group’s Third-Party Voice Mail Support service to use the phone number of the newly created hunt group as the address of the Voice Mail server. Additional configuration of the Third-Party Voice Mail Support service may be required.

THIRD-PARTY VOICE MAIL SUPPORT

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

DESCRIPTION

This service allows the forwarding of busy, unanswered, and/or received calls outside of primary zone to an external voice messaging platform. The destination can be a phone number or a URL.

This service has the lowest level of precedence, which means that Call Forwarding and Voice Messaging services have precedence over it.

Message Deposit

Calls that reach a busy or no-answer condition are redirected to the third-party messaging server configured for the group. Once the call is answered by the third-party messaging system, the call control is handed off to the messaging system for further processing.

The messaging capabilities are integrated with CommPilot Express.

Message Retrieval

The user can retrieve their voice messages by pressing the Messaging button on their phone (if available), by dialing *86 (default), or by dialing their own number from their phone. Clearspan redirects the call to the external voice mail platform.

The user can always retrieve their voice messages by calling the voice portal of the external voice mail platform.

Message Waiting Notification

Incoming message waiting indicators with the custom mailbox ID of the user are handled by the Third-Party Voice Mail service.

CONFIGURATION

The service provider administrator configures the phone number or SIP-URI of the external voice mail platform for the group and the custom mailbox ID for the user to use (as diversion header) when redirecting calls to the external platform. If the custom mailbox ID is not configured, the system uses the user’s (or group’s) E.164 phone number.

The group administrator can enable or disable the service for their group.

The user can enable or disable their own service; specify whether all, busy, and/or unanswered calls should be sent to voice mail, and select the number of rings before the call is considered as unanswered.

Note however, that the number of rings setting applies to all services with no-answer handling, that is, Voice Messaging, Third-Party Voice Mail Support, Call Forwarding No Answer, and Sequential Ringing.

Calls can also be redirected to voice mail using feature access codes. For more information, see Call Forwarding Always to Voice Mail, Call Forward Busy to Voice Mail, and Call Forward No Answer to Voice Mail.

VOICE MESSAGING

Clearspan Voice Messaging enables users to record messages from callers for calls that are not answered within a specified number of rings, for calls that receive a busy condition, or for calls received when the user is outside of their primary zone.

DESCRIPTION

Clearspan messaging 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 and to any specified mail or short message service (SMS) account (for example, cell phone)
- Integration of the messaging capabilities with Clearspan 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
- Sharing of Voice Mail service with other phones, such as a cell phone (called Voice Messaging Aliasing). With this capability, users can forward any phone to the voice portal phone number and have calls sent directly to their mailbox greeting.

The following subsections describe the service in greater detail.

Message 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.

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.

When leaving a message, the caller has access to the following functions:

- Set the message status to urgent and/or confidential
- Review the message and erase, record it again, or deposit it
- Send an unsent recorded message before being transferred to an attendant
- Choose between being transferred to an attendant or returning to the voice portal after sending the message

The access to the voice portal is configurable through the web portal or the Application Server command line interface (CLI). When it is not allowed, the related prompts are not played.

Video Support

Clearspan messaging allows for providing a video greeting to video-enabled callers, and also allows callers to leave video messages for the user.

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 in audio or video format. Users are notified that a specific message is in video format. Messages can be retrieved via e-mail through the voice portal, but only the audio portion is played back.

Storage

Messages are stored on standard e-mail servers (POP3, IMAP4, or Microsoft Exchange Server) as *.WAV* or *.MOV* 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.

Message Retrieval

The user can retrieve their voice messages by pressing the Messaging button on their phone (if available), by dialing *86 (default), or by dialing their own number from their phone. The user is then connected to their Voice Mail Retrieval menu.

When a user who does not have a Voice Messaging service configured dials *86 (default), they are redirected to the voice portal and prompted to dial a mailbox ID.

When a user without Voice Messaging service configured calls their own extension, they hear a busy tone.

Users who have their voice mail configured to be forwarded to e-mail, can retrieve their voice mail messages from their location, from a third-party location, or from any standard e-mail client.

When retrieving emails 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 (that is, 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.

Note: The reply option works only if the caller who left the message is in the same group as the user who received the voice mail. However, if the call was handled as a Public Switched Telephone Network (PSTN) call, (as is the case with Call Processing policies, such as *Force All Calls to Use the Network* or *Force All Calls to the Network Except Extension/Location*), then the reply does not work. This is because the original call was sent to the network and it was not seen as a PSTN call or a group call.

Messages are retrieved as attachments to e-mails and can be listened to with standard audio software. Messages received can be manipulated similar to any other e-mail (stored, forwarded, replied to, and so on).

Note: If the user uses the Callback service, then any transfer attempt with another party drops all calls. This behavior protects the voice mail's (VM's) user against fraud. If calls are not dropped, the transferred party may jump into the VM, which is not the desired behavior.

Message Waiting Notification

When the user receives new messages, they can be notified by standard message waiting indication mechanism (a stutter dial tone and message waiting lamp). Users can also request a notification to be e-mailed to a specific location, such as a cell phone, when a voice message is received.

The Message Summary Updates feature synchronizes the message-waiting summary after a reboot to reflect the true status of the mailbox. Since the message summary update is only done on device registration, the devices that do not register to the Application Server do not benefit from this feature.

CONFIGURATION

The system administrator configures the system Voice Messaging settings such as the address of the Voice Mail server, voice portal scope, message aging, message length, mailbox size, and default "From" e-mail address.

The service provider and enterprise administrators can override the system default setting for the *From* field used when sending an e-mail for message deposit and message notification at the service provider/enterprise level.

The group administrator configures voice messaging settings that apply to the whole group, such as message aging, default group mail server, and mailbox size, and specifies whether users are allowed to configure their own advanced mailbox settings. The group administrator also specifies whether the *Send message to Entire Group* option is allowed in voice message retrieval, compose, and forward menus.

Note: These attributes apply only to the users of the group using the default group mail server.

In addition, the group administrator assigns Voice Messaging User, and (optionally) Voice Messaging User – Video services to the user and specifies the maximum allowed size of the user's voice mailbox.

The user can:

- Enable or disable the service.
- Specify whether all, busy, and/or unanswered calls should be sent to voice mail.
- It is strongly recommended to avoid setting the number of rings before the call is considered unanswered to a value that is equivalent to the value assigned to the Maximum Duration for Unanswered Calls call processing policy. The concurrency between the two timers can cause one or the other to trigger first based on the activity on the server and the session topology. Therefore, a spacing of at least one second is recommended. The number of rings can be converted into duration with the ring period based on the country code setting.
- Specify whether call should be forwarded to voice mail when the user is outside of their primary zone.
- Specify the mode of retrieval of voice mails.
- Configure their Voice Messaging 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 sent directly to their mailbox greeting.
- Define the passcode used to retrieve messages through the CommPilot voice portal.
- Activate and deactivate message waiting indication on the phone.
- Activate and configure sending message waiting indication to an email address.
- Define distribution lists to which to send voice messages.
- If authorized, specify the server where messages are stored (personal or default mail server).
- The user can also record new greetings and a personalized name that is played as part of the default system greeting through the voice portal.

Optionally, the user can configure their service as a greeting-only service, which provides an informational message, but prevents the calling party from leaving a message in the called party's voice mailbox. Once the informational message finishes playing, the call is either disconnected or forwarded to a configured number.

VOICE PORTAL

The voice portal provides an IVR interface 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.

Users can automatically log in to the voice portal when calling from their own phone or device.

DESCRIPTION

The voice portal provides a convenient way for users to manage their services from any phone. The voice portal allows the users to:

- Log in by dialing the voice portal number or extension, or by dialing *62 (default)
- Manage their voice mailbox (see Voice Messaging):
  - Retrieve messages
  - Compose, forward, or reply to messages
  - Change greetings
- Activate, deactivate, and program their Call Forwarding Always service (See Call Forwarding Always.)
- Record a personalized name for an Auto Attendant and standard voice mail greetings
- Modify the passcode
- Record Auto Attendant greetings (group administrator only)
- Make an external call

To access the voice portal, users must dial either the number of their group voice portal or the Voice Portal feature access code.

Upon connecting to the voice portal, the user is optionally played a branding announcement, followed by a prompt for their number and passcode. (When 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 auto login.) Upon successful authentication, the user is presented with the main menu that offers the options described previously. (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.)

When auto login is enabled, all scenarios in which the system recognizes the calling user (and would normally prompt for a passcode) result in an automatic authentication, and the passcode collection phase is skipped. Examples of call scenarios with auto login enabled follow:

- Users call their service provider/enterprise or group voice portal number from their own phone.
- Users call their phone number from their phone.

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 allow users to revert to the main menu, so multiple options can be selected during the same session.

It is possible to originate an expensive call from the voice portal. The Expensive Call Notification warning announcement applies as it does for any other originating call.

Voice Portal Wizard

The Voice Portal Wizard is (optionally) assigned to groups, and assists users the first time they log in to the voice portal.

Upon logging in, users are guided through the following steps:

- Change passcode from the default passcode (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

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 rules apply each time users change their passcode, which can be done either via the web portal or the voice portal.

Passcode rules include the following:

- Passcode aging - This rule sets the maximum age for a passcode and forces the user to change their passcode before they are granted access to their voice portal.
- **Passcode length** – This rule specifies the minimum required length of the passcode.
- **Trivial passcode** – Rejects password that are considered trivial, such as:
  - Passcodes with repeated or consecutive digits
  - The user's own extension or phone number
  - The user's own extension or phone number reversed
  - Recent or reversed passcode
- Passcode Lockout – The user is locked out of voice portal access after a configurable number of unsuccessful login attempts in a row. When this occurs, 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. The user's voice portal account must be reset by the group administrator via the user's *Passcode Reset* page before it can be used again.

Voice Portal Scope

The voice portal scope can be set to "service provider/enterprise" or "group".

When the scope is set to "service provider/enterprise", voice portals within the same service provider/enterprise cooperate. To log in, the user can dial the phone number of any voice portal within the service provider and is silently redirected to the user's actual group voice portal.

When the voice portal scope is set to "group", voice portals do not cooperate with any other voice portal in the system.

Users can be identified by their extension (within the same group) or their phone number. In addition, enterprise users can be identified by location code + extension. This applies when logging in to the voice portal, when sending, forwarding, or replying to messages, and when creating distribution lists.

When the voice portal scope is set to "enterprise", location code + extension can be used as the redirection mechanism to refer to a voice portal (instead of a direct inward number) if no direct inward number that can be dialed is assigned to a voice portal.

System and Network Voice Portal

The system administrator can create voice portals at the system level.

The system voice portal provides the same functionality as does the group voice portal with these advantages:

- A single voice portal instance created at the system level does the work that is done by multiple group voice portals in the current system-wide voice portal approach. This results in a reduction in the number of phone numbers and memory resources consumed by multiple group voice portals.
- When the system voice portal is part of a network voice portal configuration, it has the ability to identify login IDs beyond the cluster where it resides.
- The network voice portal allows the use of a single voice portal number accessible by all subscribers in the network. The network voice portal calls are redirected to the appropriate system voice portal. The system voice portal is not directly dialable when it is configured as part of the network voice portal solution.

The system voice portal and the group voice portal can co-exist in the same system but they do not interact with each other. Users who access the system voice portal hear the system voice portal greetings.

Residential Voice Portal

This feature allows service providers to create a voice portal that spans all service provider groups 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.

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.

The Residential Voice Portal feature removes the requirement of a full national number (DN) per group by allowing phantom numbers to be used to identify a group voice portal. Only one of the voice portals within the scope must have a DID. The redirection involves using the Clearspan user ID of the voice portal in such a case. Note that in this case, the group should have a group CLID set.

Voice Portal Calling

Voice Portal Calling allows an authenticated user to originate calls from the voice portal. This service is particularly useful for traveling users who 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 are eventually charged against their account. For similar reasons, this feature can be useful for the employee working at home who 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 Outgoing Calling

Plan, Account/Authorization Code, and voice Virtual Private Network apply on the outgoing calls made from the voice portal. This also means that accounting records are 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 they can "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 keys that can be used to navigate through the voice portal menus.

For each menu and submenu of the voice portal, the association of keys with actions (choices of each menu) is configurable. The following 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 they are reserved for non-configurable purposes.

The key is either one digit (0 through 9), *, or # (or "none" when choosing to disable an optional menu option), except for the prompt to initiate a new call when using Voice Portal Calling (currently set to "##"), in which case the selection is made of a sequence of two to three keys, where the inter-digit time-out 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 through 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

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 voice portal keys also requires a rerecording of these new announcements.

Extended Availability of Options

Some submenu options are offered from more than one menu. For example, the *reply to a voice message* option is 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 of these menus.

Not all options are available from any menu; only the relevant options are presented in each menu. Without allowing complete reorganization and dynamic definitions of new menus, this allows for some flexibility on how menus are structured.

CONFIGURATION

The system administrator can customize the voice portal menus.

In the centralized (Amplify) architecture, the system administrator cans also allow or disallow voice portal menu customization for an enterprise or service provider group. When allowed, enterprise and service provider group administrators can customize voice portal menus for their organizations.

The group administrator configures the group voice portal through the CommPilot web portal. The voice portal configuration includes the following:

- Activating/deactivating the voice portal.
- Basic information, such as name, calling line identity, language, and time zone
- Phone number and/or extension.
- Assigning an administrator passcode to the voice portal.
- Activating/deactivating the voice portal wizard.
- Allowing voice portal login using phone numbers or voice messaging aliases (in addition to an extension).

No configuration is required for the Residential Voice Portal feature because the user ID is automatically created when assigning the Voice Messaging Group service to a group.

The system administrator configures the system voice portal through the CommPilot web portal. The parameters that can be configured include the following:

- Basic information, such as name, calling line identity, language, and time zone
- Phone number
- Network voice portal number, if applicable
- Voice portal and messaging greetings

PROVISIONING AND CONFIGURATION TOOLS

The following capabilities are mainly intended for service provider and enterprise administrators to help with provisioning and configuration tasks for their organizations. Some capabilities are also available to group administrators and/or users.

ALTERNATE USER IDS

Alternate User IDs allow a carrier to define an alternate identification that the subscribers can use to log in to Clearspan.

DESCRIPTION

Some carriers deploy Hosted Private Branch Exchange (hPBX) service without an end-user portal or they use a portal or other clients where subscribers do not have to identify themselves using the Clearspan user ID. In such cases, carriers often set up the subscribers' Clearspan user IDs with phone numbers, account numbers, and other long identifiers.

Now that many of these carriers are transitioning from hosted PBX to Unified Communications, which requires deployment of clients such as Business Communicator, subscribers need to identify themselves with Clearspan.

As Clearspan user ID, is not always easy for subscribers to remember, this feature allows a carrier to define an alternate user identification that the subscribers can use to log in to Clearspan. Typically, this user ID is in an email format, for example, joe@carrier.com, but it can be a simple name, such as "joe".

A user can have up to eight alternate user IDs.

CONFIGURATION

Only the system administrator can configure alternate IDs for the users.

BUSINESS TRUNKING

Business Trunking allows for licensing a maximum number of simultaneous calls that a specific group of users can have. It uses the concept of a trunk group to serve PBX-type customer premises equipment (CPE). This framework provides better support for intelligent CPE such as PBXs, while still allowing Clearspan services to be offered to CPE users.

DESCRIPTION

The Business Trunking framework enhances the Clearspan data model by using the concept of a trunk group to serve PBX-type customer premises equipment sharing the following characteristics:

- The CPE provides most of the user services, which do not need to be provided by Clearspan.
- The number of users on the CPE may be greater than the number of users on Clearspan. As a result, the number of users on Clearspan does not provide a reliable indication of the system resources consumed by the CPE. (For example, a single Clearspan user may be mapped to an Auto Attendant on the CPE, which covers an unlimited number of CPE users).
- The CPE can pull some services from Clearspan, such as:
  - Account/Authorization Codes
  - Calling Plans
  - Auto Attendant
  - Hunt Groups
  - Call Centers
  - Series Completion
  - Meet-Me Conferencing
  - Messaging
  - Find-me/Follow-me
- The CPE can pull network services from Clearspan, such as:
  - Abbreviated Dialing
  - Voice VPN
  - Far-End Hop-Off

As a result, to serve this CPE, Business Trunking provides licensing of call bandwidth independently of the number of users while allowing Clearspan services to be assigned to CPE users as required.

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 to the group as usual. In addition, one or more trunk group(s) is assigned to the group with the following characteristics:

- **Call capacity management** – This includes the maximum number of concurrent calls (and optionally the maximum bursting number of concurrent calls) that trunk group users are allowed and the treatment to apply to calls when call capacity is exceeded. Call capacity can also be configured independently for incoming and outgoing calls.

- **Trunk users** – These users are mapped to the CPE, using the trunk as an access method. All calls processed for these users are accounted for by the trunk group call capacity.

  In addition, regular Clearspan users can also be included in a trunk group for call capacity management.

- **Trunking device** – This is the access device that represents the CPE (and that holds the IP address and port information) and the treatment to apply when the device becomes unreachable. There can only be one trunking device per trunk group, so all trunk users make use of that trunking device. Any group device can be used for the trunk. Shared devices for a service provider/enterprise can also be used.

- **Trunk group identity** – This includes various parameters that specify how trunk group identity is used in call processing.

In addition, a pilot user can be assigned to a trunk group.

CONFIGURATION

- **Call capacity allocation** – The system provider allocates trunking calls (and bursting trunking calls) to a service provider and the service provider then distributes these 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 creation** – The trunk group is a virtual user and its configuration is similar to other virtual users (Auto Attendant, Call Center, and so on). Items specific to the trunk group include the device associated with it, the trunk group identity management options, the call capacity management attributes, the list of group users associated with the trunk group for call capacity management, the authentication password, and a pilot user.

- **Trunk group users' creation** – Users can be added to a trunk group either individually by associating the trunk group with the user's profile as if it were a device or in bulk by using the trunk group user creation wizard.

CALL PROCESSING POLICIES

This feature provides explicit control of certain Clearspan call processing behavior.

DESCRIPTION

Call Processing policies allow administrators to specify the behavior to apply to calls.

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 they default to the system-wide defaults.

The system selects the policy based on the level at which it is configured on a per-call basis. Therefore, the user policy is used if it is configured and enabled. If it is not configured and enabled, then the group policy is used if it is configured and enabled. If it is not configured and enabled, the service provider/enterprise policy is used, if it is configured and enabled. If none of these policies is enabled, the policies default to the system-wide settings.

The policies in the following subsections can be configured. If a policy can only be defined at certain levels, the levels are specified.

Calling Line ID

These settings allow the administrator to specify the type of identification that should be used for emergency, non-emergency, and redirecting calls, and whether to block the calling name ID for redirecting calls.

In addition, the use of group/department ID as calling line identity can be enabled or disabled at group and user levels.

Media Policies

Fax machines and modems require the use of a clear channel and an uncompressed codec. This setting allows an administrator to force a user's device to use an uncompressed codec. Alternatively, if provisioned, an alternative supported medium can be selected from the list provided.

Call Limits

These policies allow the administrator to impose call limits on various categories of calls, for example, the maximum number of concurrent calls of certain types or the maximum duration of certain types of calls.

Note that these policies only count originating and terminating calls for a user. Redirected calls are not counted.

If a user exceeds the maximum allowed number of concurrent calls, the calling party is provided busy treatment. If the terminating user has no active services that trigger on busy assigned, busy tone is provided to the calling party. If a user exceeds the maximum time on a given call, the call is torn down and calling party is given reorder tone.

Note that the Maximum Duration for Unanswered Calls policy interacts with all the services with no-answer handling, such as Voice Mail, Third-Party Voice Mail Support, Call Forward no Answer, and Sequential Ringing. It is strongly recommended to avoid setting that policy to a value that is equivalent to the number of rings before no-answer processing. The concurrency between the two timers can cause one or the other to trigger first based on the activity on the server and the session topology. Therefore, a spacing of at least one second is recommended. The number of rings can be converted into duration with the ring period based on the country code setting.

Conferencing

This policy specifies the conference SIP-URI to use to establish conference calls (using Three-Way or N-Way Conferencing). This policy is only defined at the service provider/enterprise level.

Translation and Routing

These settings allow the administrator to specify the Translation and Routing policies to apply to calls. They can be defined at the service provider/enterprise and group levels.

Translation and Routing policies allow the administrator to force all calls to route via the network interface, apply calling line identity restrictions to calls, and enable or disable call typing.

Note that a Network Server with a private dial plan is required for extension dialed and location code/extension dialed calls to work properly when forced to the network.

In the enterprise model, extension dialing can be allowed or disallowed for users in different enterprise groups.

Dialable Caller ID

This policy allows the administrator to enable or disable the delivery of the calling number in dialable format. The use of a dialable caller ID allows one to make a distinction between local and toll inbound calls based on the calling number, and calls can be returned directly from the local phone call logs without having to manipulate the digits.

Phone List Lookup

This policy allows the administrator to decide whether Clearspan should try to determine the calling name for incoming calls that do not include calling name information. When the Phone List Lookup policy is enabled, this is accomplished by attempting to find the calling number in the called user's personal directory, called user group's Common Phone List, or called user enterprise's Common Phone List. If the calling number is located in one of these lists, then the calling name provisioned for it is used and presented to the called party.

CONFIGURATION

Call Processing policies can be set at the service provider, group, and user levels. For more information, see *Description*.

CLASSMARK

This service allows a classmark to be assigned to a user and communicated within SIP messaging between Clearspan and the PSTN during call setup.

DESCRIPTION

Classmark is an origination service, which allows the users' and virtual users' (for example Meet-Me Conferencing) classmarks to be sent to the PSTN.

In addition, the system can be configured to allow a classmark received from the PSTN to be proxied toward the PSTN.

The following are some of the redirection services in which the classmark may be proxied to the PSTN:

- Call Forwarding Always
- Call Forwarding Busy
- Call Forwarding No Answer
- Call Forwarding Not Reachable
- Call Forwarding Selective
- Blind Call Transfer
- Clearspan Anywhere
- Remote Office
- Sequential Ringing
- Simultaneous Ringing

The classmark cannot be proxied for Call Transfer with Consultation since the call with the transfer-to party has already been started or established prior to the transfer.

CONFIGURATION

Classmarks are configured system-wide by the system administrator.

Service providers can then associate a classmark with the user. Group administrators and users have no access to the configuration of this service.

COMMUNICATION BARRING – FIXED

The Communication Barring – Fixed (CBF) service provides a mechanism to impose restrictions on the type and duration of calls that subscribers can make at a given time.

DESCRIPTION

This service allows a system administrator to specify various criteria to apply to calls and the actions (such as, block, allow, or transfer) to take for calls that satisfy the specified criteria. Different rules can be applied to originating, redirecting, incoming, and Call Me Now calls. These criteria are combined into communication barring profiles.

Communication barring profiles are used in Network Classes of Service (NCOS) assigned to users to specify the types of calls users can make.

Authorization Code NCOS Capability

This feature adds flexibility to the Clearspan Communication Barring service, enabling Clearspan to apply a Network Class of Service (NCOS) dynamically for originating calls. Clearspan can prompt the originating user for a code, and then select and apply an NCOS based on the code entered.

To enable the new capability, an administrator creates new Communication Barring authorization codes at the enterprise or group level and assigns these codes to an NCOS. If an originating user enters one of these authorization codes (after being prompted by a Communication Barring rule), the Application Server selects and applies the assigned NCOS.

The new scenario requires the Application Server to enforce Communication Barring twice. First, the Application Server enforces an initial Communication Barring profile selected by the NCOS assigned to the originating user. When the Application Server enforces that Communication Barring profile, it prompts the originating user to enter an authorization code. Then, based on the code entered, the Application Server selects a secondary NCOS and enforces the active Communication Barring profile for that NCOS.

If a group or enterprise authorization code does not have an NCOS assigned, then the Application Server allows the call.

CONFIGURATION

The system administrator configures communication barring profiles and uses them to create Network Classes of Service.

The following types of criteria can be used for originating, redirecting, and Call-Me-Now calls:

- Time criteria
  - Holiday schedule
  - Time schedule
- Network and call type criteria
  - Charge indicator
  - Type of network
  - Category
  - Network indicator
  - Configurable call types
  - Alternate call indicators
  - Virtual On-Net call types

In addition, criteria based on digit patterns can be defined and used for incoming calls.

Enterprise administrators can create authorization code and assign a Network Class of Service to it. Group administrators can assign a Network Class of Service to a group-level authorization code.

WEB PORTALS

web portals allow administrators at different organizational levels to provision and configure resources and services for their organization. It also allows users to self-manage their services.

Clearspan provides flexibility in managing service configuration using web portals. The web portals (called web portals) and service management permissions provide secure access to the information for each user type or role: system provider, service provider, enterprise administrator, group administrator, department administrator, and user. For more information about service management permissions, see Restricted Administration Access.

Clearspan provides a different web portal or interface for each role:

- The system provider administrator uses the web portal to establish system-wide settings, such as setting messages callers hear when they interact with Clearspan services.
- Service provider and enterprise administrators use the web portal to authorize specific services and resources for groups and configure organization-wide settings.
- Group administrators use the group version of the web portal to manage group resources and activities and provision services for users.
- Department administrators use the department administrator version of the web portal to perform a limited number of functions related to their department.

CONFIGURABLE DEFAULT FEATURE ACCESS CODES

This feature provides a system-level configurable default FAC set. When a new service provider or enterprise is created, their FAC table is initialized based on the system defaults. In addition, when new groups are created under a service provider or enterprise, their FAC table is initialized based on the service provider's defaults.

The feature also provides a configurable default Speed Dial 100 prefix at the system level, which is used as default on authorization of Speed Dial 100 service to groups.

DESCRIPTION

This feature allows the configuration of the default main and alternate feature access codes at the system, service provider/enterprise, and group levels. It also allows the system administrator to view all feature access codes that are available (but not necessary licensed) on the system. Service provider/enterprise and group administrators only see the feature access codes that were assigned to their organization.

The viewed data of each FAC contains:

- Feature access code name
- Main FAC
- Alternate FAC
- Default Speed Dial 100 Prefix

This feature allows system administrators to predefine default FACs with values of their choice, so that all newly provisioned service providers and enterprises automatically inherit feature access codes defined by system administrator. Service providers can assign the default FAC and any group that is created under that service provider is assigned the default FAC that was created by the service provider. The service provider or group administrator still has the privilege to change the assigned default FACs to any other allowed values.

This feature does not change how feature access codes work. It only allows for changing how the codes are used to invoke features.

Modified feature access codes are applicable in newly created (after modification) service providers, enterprises, and groups. The feature access codes in service providers or groups that were created prior to the default feature access codes modification are not affected.

The default Speed Dial 100 prefix can also be modified. When Speed Dial 100 service is authorized to a group, the group inherits the configured value of the Speed Dial 100 prefix.

Groups that had the Speed Dial 100 service authorized prior to the default Speed Dial 100 prefix modification are not affected, and the original Speed Dial 100 prefix is used. The group administrator can change their Speed Dial 100 prefix if desired. In an IMS deployment, the Speed Dial 100 prefix can be empty. However, caution should be used when creating the Speed Dial 100 prefix to avoid conflict with feature access codes, extensions, and emergency numbers.

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 the event of collisions, the newly authorized FAC (to the group) is left blank and a warning message is generated.

CONFIGURATION

System and service provider/enterprise administrators can create default FAC values for various services. These FACs are assigned to service providers and groups respectively as default FACs for authorized services when service providers or groups are created. Service provider and group administrators can then change the default values if desired.

The system default FAC values and Speed Dial 100 prefixes are configured using the CLI. The service provider and group FAC values are configured using the web portal.

The group *FAC Configuration* page provides a reset link used to reset the group FAC values back to the default service provider configured values.

System Default Feature Access Codes

The following table shows the system default feature access codes.

CODE	FEATURE
*34	Advice of Charge Activation
*77	Anonymous Call Rejection Activation
*87	Anonymous Call Rejection Deactivation

CODE	FEATURE
52	Anonymous Call Rejection Interrogation
#8	Automatic Callback Deactivation
#9	Automatic Callback Menu Access
*14	Clearspan Anywhere E.164 Dialing
*15	Call Bridge
*72	Call Forwarding Always Activation
*73	Call Forwarding Always Deactivation
21	Call Forwarding Always Interrogation
*21	Call Forwarding Always To Voice Mail Activation
#21	Call Forwarding Always To Voice Mail Deactivation
*90	Call Forwarding Busy Activation
*91	Call Forwarding Busy Deactivation
67	Call Forwarding Busy Interrogation
*40	Call Forwarding Busy To Voice Mail Activation
#40	Call Forwarding Busy To Voice Mail Deactivation
*92	Call Forwarding No Answer Activation
*93	Call Forwarding No Answer Deactivation
61	Call Forwarding No Answer Interrogation
*41	Call Forwarding No Answer To Voice Mail Activation
#41	Call Forwarding No Answer To Voice Mail Deactivation
*94	Call Forwarding Not Reachable Activation
*95	Call Forwarding Not Reachable Deactivation
63	Call Forwarding Not Reachable Interrogation
#76	Call Forwarding Selective Activation
#77	Call Forwarding Selective Deactivation
54	Call Forwarding Selective Interrogation
*67	Calling Line ID Delivery Blocking per Call

CODE	FEATURE
*31	Calling Line ID Delivery Blocking Persistent Activation
#31	Calling Line ID Delivery Blocking Persistent Deactivation
*65	Calling Line ID Delivery per Call
*68	Call Park
*88	Call Park Retrieve
*98	Call Pickup
*48	Call Recording – Pause
*49	Call Recording – Resume
*44	Call Recording – Start
*45	Call Recording – Stop
*11	Call Retrieve
*69	Call Return
#92#	Call Return Number Deletion
53	Call Waiting Interrogation
*43	Call Waiting Persistent Activation
#43	Call Waiting Persistent Deactivation
*70	Cancel Call Waiting
*99	Clear Voice Message Waiting Indicator
33	Communication Barring User-Control Activation
#33*	Communication Barring User-Control Deactivation
*#33#	Communication Barring User-Control Query
56	Connected Line Identification Restriction Interrogation
*57	Customer Originated Trace
*97	Directed Call Pickup
*33	Directed Call Pickup with Barge-in
*55	Direct Voice Mail Transfer
*80	Diversion Inhibitor
*78	Do Not Disturb Activation

CODE	FEATURE
*79	Do Not Disturb Deactivation
#83	Escalate Call to Supervisor
#63	Executive-Assistant Call Push
#64	Executive-Assistant Initiate Call
#65	Executive-Assistant Opt-in
#66	Executive-Assistant Opt-out
#61	Executive Call Filtering Activation
#61	Executive Call Filtering Deactivation
*26	Find-me/Follow-me Call Push
*22	Flash Call Hold
#72	Forced Forwarding Activation
#73	Forced Forwarding Deactivation
#58	Group Call Park
#51	Hunt Group Busy Activation
#52	Hunt Group Busy Deactivation
#53	Hunt Group Busy Interrogation
#82	Initiate Silent Monitoring
*66	Last Number Redial
#96	Legacy Automatic Callback Cancellation
*96	Legacy Automatic Callback Invocation
*12	Location Control Activation
*13	Location Control Deactivation
#80	Make Outgoing Call as Call Center
#81	Make Personal Outgoing Call
#23	Mobility Call Anchoring Activation
*23	Mobility Call Anchoring Activation Per Call
#24	Mobility Call Anchoring Deactivation
*24	Mobility Call Anchoring Deactivation Per Call

CODE	FEATURE
#29	Mobility Calling Line ID Activation
*28	Mobility Calling Line ID Activation Per Call
#28	Mobility Calling Line ID Deactivation
*29	Mobility Calling Line ID Deactivation Per Call
#84	Monitoring Next Call
*60	Music On Hold Per-Call Deactivation
#70	Night Service Activation Manual Override
#71	Night Service Deactivation Manual Override
*610	No Answer Timer
*84	Number Portability Announcement Activation
*85	Number Portability Announcement Deactivation
*71	Per-Call Account Code
*50	Push To Talk
51	Selective Call Rejection Interrogation
*75	Speed Dial 100
*74	Speed Dial 8
*47	Sustained Authorization Code Activation (calls unlocking)
*37	Sustained Authorization Code Deactivation (calls locking)
*86	Voice Mail Retrieval
*62	Voice Portal Access

ENTERPRISE TRUNKING

An enterprise trunk combines multiple trunk groups and allows the trunk groups to work together to provide enhanced business trunking capabilities for large enterprises. It can be created for an enterprise or service provider group.

DESCRIPTION

Subscribers can be assigned to an enterprise trunk, rather than to a single trunk group. These subscribers can originate or terminate calls through any of the trunk groups assigned to the enterprise trunk. Thus, an enterprise trunk provides routing flexibility and redundancy that are not possible with a single trunk group.

An enterprise trunk permits originations and terminations via multiple physical paths, thus allowing for redundancy, load balancing, far-end hop-off, and so on. **Error! Reference source not found.** shows a diagram of a possible Enterprise Trunking deployment.

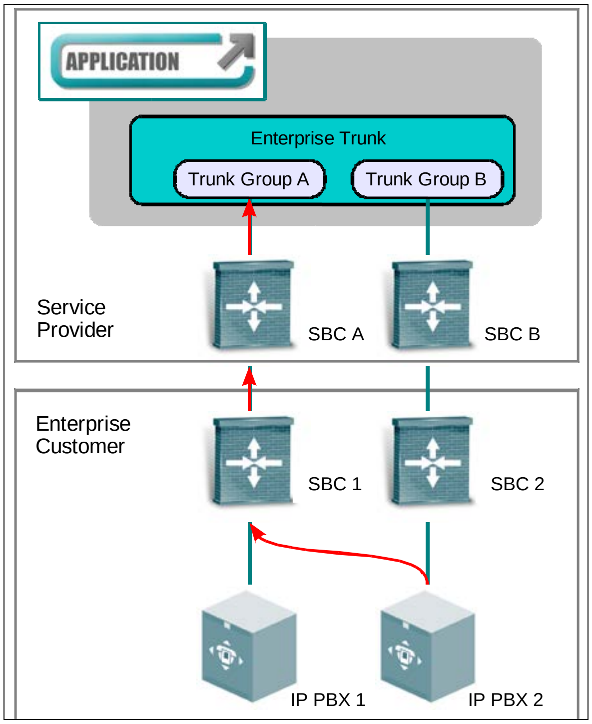

Figure 7 Example of Enterprise Trunking Deployment

CONFIGURATION

Enterprise trunks are created by enterprise administrators and group enterprise trunks are created by group administrators.

The Trunk Group service must be assigned to the enterprise or group before enterprise trunks can be created. In addition, trunking call capacity must be configured before enterprise trunks can be created.

To create an enterprise trunk, follow these steps:

1. Select the routing policy.
2. Select trunks groups and assign routing policies to trunk groups.
3. Assign users to the enterprise trunk.

HIERARCHICAL COMMUNICATION BARRING

Hierarchical Communication Barring (HCB) enables the service provider or enterprise administrator to apply calling restrictions to their subscribers, in addition to the restrictions imposed at the system level using the Communication Barring – Fixed service.

DESCRIPTION

Hierarchical Communication Barring allows the service provider and enterprise administrators to create communication barring profiles at the service provider/enterprise level and apply them to their groups. They can also select communication barring profiles that can be assigned to users in a group. A group administrator with appropriate permissions can then assign communication barring profiles to individual users.

Communication Barring – Fixed rules have precedence over the Hierarchical Communication Barring rules. If the call attempt is blocked by a system-level rule, the service provider/enterprise rules are not examined.

CONFIGURATION

The system administrator can assign system-level communication barring criteria to service providers and enterprises for use in communication barring profiles.

Service provider and enterprise administrators can define additional criteria based on digit patterns.

Using both the system-level and the service provider/enterprise-level criteria, administrators define communication barring profiles and assign them groups.

Enterprise administrators can assign user-assignable profiles to groups, and group administrators can assign those profiles to users.

LARGE ENTERPRISE SUPPORT

Clearspan provides support for large multisite enterprises, with the enterprise layer in the Clearspan provisioning model. This layer allows the Clearspan customer to better model, administer, and manage large multisite enterprises.

An enterprise is a specific type of service provider. It has all the capabilities of the service providers in addition to enterprise-specific capabilities. An enterprise should be used when a company has multiple sites or heavily geographically distributed users.

ENTERPRISE PRIVATE DIALING

This capability allows for creating a private dialing plan shared between the multiple sites making up an enterprise. The enterprise dialing plan allows users of the enterprise to call one another using location codes and extensions instead of full phone numbers.

When creating the sites (groups), the administrator can assign location codes to each group, which can be used by enterprise users to make calls between sites, using a private dialing plan and the previously configured VPN policies.

All the enterprise private dialing changes and policies carried out by the administrator on the Application Server are automatically synchronized on the Network Server without requiring the administrator intervention.

It should be noted however, that the Network Server can be accessed directly by the enterprise administrator to provision the enterprise routing policies or to configure advanced routing policies that are not exposed via the enterprise administrator portal on the Application Server.

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 the enterprise or 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.

A group administrator can extend the enterprise department hierarchy, but cannot create departments at the enterprise level.

All the departments that belong to a group must have a unique name within that group. Similarly, 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. However, users cannot belong to a department that belongs exclusively to another group.

In addition, it is not possible to create department administrators for departments defined at the enterprise level.

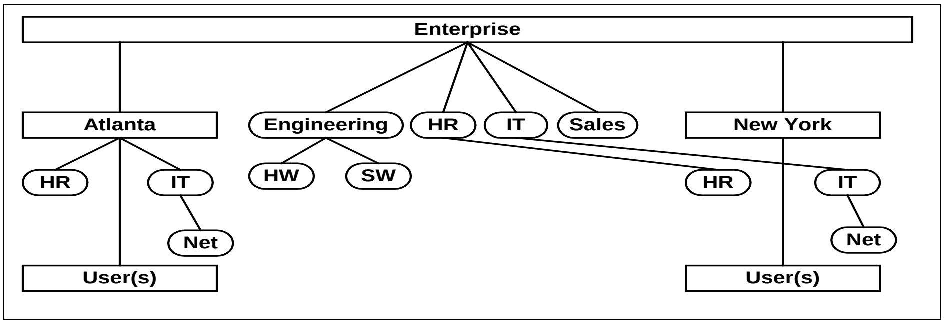

Figure 8 Departments

The following table lists the departments shown in the figure above.

DEPARTMENT	DESCRIPTION
Engineering	This may contain any user.
Engineering\HW	This may contain any user.
Engineering\SW	This may contain any user.
HR	This may contain any user.
HR(Atlanta)	This may contain Atlanta users only.
HR\HR(New York)	This may contain New York users only.
IT	This may contain any user.
IT(Atlanta)	This may contain Atlanta users only.
IT(Atlanta)\Net	This may contain Atlanta users only.
IT\IT(New York)	This may contain New York users only.
IT\IT(New York)\Net	This may contain New York users only.
Sales	This may contain any user.

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.

ENTERPRISE-WIDE GROUP SERVICES

The Clearspan 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 who belong 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
- Call Center settings

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
- Hoteling
- Push To Talk
- Voice Messaging
- Voice Portal
- Virtual On-Net Enterprise Extensions

These enhancements allow these services to be used between users who belong to different groups in the enterprise.

Enterprise Voice Portal and Messaging – For more information, see Voice Portal Scope.

ENTERPRISE DIRECTORY

The enterprise directory contains a list of all the assigned phone numbers in the enterprise. It includes users, Auto Attendants, hunt groups, call centers, Meet-Me conferences, and voice portal numbers. Each entry in the directory contains the name of the entity with their DN, extension, group, and department. The enterprise directory can be viewed by all the users in the enterprise.

The enterprise directory also shows the common phone numbers and Virtual On-Net users defined by the enterprise administrator.

NETWORK CLASSES OF SERVICE

This feature provides a mechanism for imposing restrictions on the type and duration of calls that subscribers can make at a given time. It allows system administrators to define Network Classes of Service that can be assigned to subscribers.

DESCRIPTION

A Network Class of Service determines the types of calls that a user is allowed to make or receive. Network Classes of Service (NCOS) are defined at the system level and consist of a set of communication barring rules, defined using the Communication Barring – Fixed service. One of the profiles is specified as a primary or active profile.

To enable NCOSs to be assigned to users and Meet-Me conference bridges, the system administrator must assign one or more NCOSs to the service providers and enterprises, which in turn assign them to their groups.

The group administrator assigns NCOS to users based on the calling capabilities they want the users to have.

CONFIGURATION

The system administrator defines Network Classes of Service by selecting one or more communication barring profiles to include in the NCOS.

A Network Class of Service is then assigned to a service provider or enterprise, as required. The service provider or enterprise administrator assigns one or more NCOSs to its groups. The group administrator assigns one NCOS to a user, that is, a user is associated with one NCOS.

In addition, one or more communication barring authorization codes may need to be created for the user or the user's group, to enable the use of certain calling features.

RESTRICTED ADMINISTRATIVE ACCESS

This feature provides granularity to the authorization of each administrative level of the web portal. Clearspan supports a "multi-tier" authorization scheme including the following: system administrator, provisioning administrator, service provider administrator, group administrator, department administrator, and user. It is possible to set a specific level of control for each administrator or user. For example, a group administrator can be created without the ability to add or remove users.

SYSTEM-LEVEL POLICIES – PASSWORD RULES

The system administrator defines rules of creating and updating passwords at the system level and can:

- Apply the rules to all administrators and users in the system.
- Apply the rules to system, provisioning, service provider, and enterprise administrators and allow service provider and enterprise administrators to create password rules for their organizations.
- Apply the rules to only to system and provisioning administrators and allow other administrators and users to use external authentication. When this option is selected, pages used to manage user and administrator passwords are modified or hidden. An additional option allows the users and service provider/enterprise/group administrators to be created either on the web portal or outside Clearspan. When the outside option is chosen, pages used to add administrators and users are also hidden. In addition, the step to add an administrator is skipped in the *Add Service Provider* and *Add Group* wizards. System and provisioning administrators are not affected.
- When the service provider and enterprise administrators are allowed to create password rules for their organizations, they in turn can either apply the rules to the entire organization, apply the rules only to group administrators, allow group administrators to create password rules for their organizations, or allow external authentication to be used for group administrators and users.

PROVISIONING ADMINISTRATION-LEVEL POLICIES

Read-only System or Provisioning Administrators

Read-only administrators at the system level view all the web pages; however, they are supplied with a modified read-only form that contains a Cancel button only.

Group Web Policies

Group-level web policies determine what and how a group can administer itself. All administrators and users who belong to the group are restricted by these policies.

These policies determine access to such capabilities as calling plans, call logs, extension dialing, and department administrators’ and users’ access to certain features and information.

Administrator Policies

When an administrator is created, the default Administrator policy is applied. It controls access to users, departments, numbers, devices, services, administrators, and the organization profile.

Service Provider, Enterprise, and Group Administrator policies can be modified on the *Administrator Modify* page for a selected administrator. The following access policy values can be assigned:

- **Full Access** – Read-write access to a page or group of functions
- **Read-only Access** – Display-only access to a page or group of functions
- **Restricted** – Either Read-only access to a page or no access to a page or group of functions
- **No Access** – No access to one function or a group of functions, for example, user functions

Not all policy values apply to all policies.

POLICY DEFAULTS

When creating administrators at a given level, a system default policy that can be preconfigured for that administrative level is applied. The exception to the default policies is made when administrators at a given level create administrators at the same level, since an administrator with limited access cannot create an administrator who has more access than they have today.

For example, 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. When a group administrator modifies another group administrator, the buttons that provide access to privileges greater than that of the “creating” administrator are disabled.

CONFIGURATION

Administrators create other administrators and modify their policies as described in the preceding section.

SCHEDULES

The Schedules feature allows administrators and users to define time and holiday schedules, which can then be used to configure services that allow selective processing based on time.

DESCRIPTION

This feature allows users and system, service provider, enterprise, and group administrators to configure time and holiday schedules. Schedules are used to configure services, such as Communication Barring, Clearspan Anywhere, or Call Forwarding Selective, which can process calls differently based on specified time criteria.

Time schedules are typically used to define business hours, meetings that recur regularly at the same time, and so on. Holiday schedules are typically used to define holidays, vacations, and special events, such as off-site meetings or conferences.

A schedule usually contains one or more events that specify when the schedule applies.

A schedule usually contains one or more events that specify when the schedule applies.

Note: By default, an empty time schedule, that is a time schedule with no events, is considered by the system as always applying. An empty holiday schedule can be considered by the system either as always applying or as never applying depending on how the system administrator has configured it. Therefore, it is recommended to test the behavior of empty holiday schedules before using them.

- Schedules defined at the system level can only be used by a system administrator.
- Schedules defined at the service provider level can only be used by the service provider administrator.
- Schedules defined at the enterprise level can be used by the users, group administrators, and enterprise administrators in the enterprise, but can only be modified by an enterprise administrator.
- Schedules defined at the group level can be used by group administrators and users in the group, but can only be modified by a group administrator.
- Schedules defined by a user can only be used to configure services for that user.

CONFIGURATION

Each user or administrator defines schedules at their administrative level.

SELECTIVE CRITERIA

The Selective Criteria feature allows a user to specify selective criteria to be used to screen calls in services that allow selective processing. It is not a user service strictly speaking but rather a tool used to configure other services.

DESCRIPTION

Selective criteria are used to screen calls in selective services, such as Call Forwarding Selective or Clearspan Anywhere, and allow those services to perform differently depending on the result of the screening. For example, Call Forwarding Selective can be used to forward an incoming call to a specified destination when the call matches a set of selective criteria specified by the user.

Selective criteria are based on the identity of the remote party, digit patterns in phone numbers, and time and holiday schedules. They are combined into criteria entries, for example, incoming calls from this number, within business hours, and during the workweek.

Multiple criteria entries can be defined for a service and a different action can be assigned to different criteria entries. For example, using Call Forwarding Selective, a user can forward calls to different destinations depending on the criteria entry that they satisfy.

The actions to be performed on calls vary from service to service. However, each selective service has a default action. For example, for the Selective Call Acceptance service, the default action is to accept calls. The default action is automatically selected when a new criteria entry is created. The user can change the action to the opposite of the default action for each criteria entry. Therefore, in the preceding example, the user can decide to accept all calls that satisfy the criteria in an entry or to reject all calls that satisfy the criteria in the entry.

The following are additional examples of services that use selective criteria to screen calls:

Call Me Now

Call Notify

Custom Ringback User

Pre-alerting Announcement

Priority Alert

Selective Call Rejection

Sequential Ringing

Simultaneous Ringing Personal

APPLY SELECTIVE CRITERIA

Selective criteria in an entry are used to determine whether the entry applies to the call and whether the action associated with the entry should be performed on the call. This section explains how selective criteria entries are applied to calls.

By default, a selective criteria entry is set to match all calls, which means that if no changes are made to selective criteria when a new criteria entry is defined, the entry always applies.

When a call is processed for a selective service, the criteria entries defined for the service, are applied to the call as follows:

- If more than one criteria entry is defined for the service, the entries are applied in order and the first entry that the call satisfies is used. The remaining entries are not considered once a match is found.
- When a matching entry is found, the action configured for that entry is performed.
- All the criteria in an entry must be met for the entry to apply. In particular, if both a time schedule and a holiday schedule are specified in an entry, the entry applies only if the time of the call meets both schedules. However, that behavior can be overridden via a system parameter and set to a logical “or” between the two schedules.
- If none of the criteria entries defined for the service applies to the call, then the opposite action for that service is performed on the call. For example, if none of the criteria entries defined for Selective Call Acceptance applies to a call, the call is rejected.

The order in which criteria entries are considered follows two rules:

- Entries in which the opposite action is selected are considered before any of the entries where the default action is selected.
- Within each action type (default or opposite), the entries are ordered and applied to calls in alphabetic order.

Note that the criteria entries defined for a given service are listed on the web portal (on the page for that service) in the sequence in which they are applied to calls.

Note: Most services cannot be turned on when no active criteria entries are defined for the service, but there are exceptions. For example, Simultaneous Ringing Personal rings secondary locations for all calls when there are no active criteria entries. For information about a specific service, refer to the section for that service.

CONFIGURATION

The system administrator can set the behavior of applying time and holiday schedules to calls when both appear in the same criteria entry, either to the logical "and" between the two schedules (default) or to the logical "or" between the two schedules via the CLI.

SESSION ADMISSION CONTROL

The Session Admission Control feature provides a mechanism to control the number of simultaneous sessions at the subscriber and logical link level for a group. The logical link capacity is controlled by creating sets of access devices logically grouped together, called session admission control (SAC) groups, and imposing restrictions on the number of concurrent sessions for a given set of devices (SAC group).

DESCRIPTION

There are many instances in which a carrier wants to limit the number of concurrent sessions going through a link (for example, due to bandwidth management). This feature provides a means to control new session creation.

The controlling plane is separated in two distinct levels:

- The first level is the *subscriber level*. An attribute that represent the maximum number of concurrent sessions can be set at the service provider, enterprise, and group levels. Once the maximum is reached, new session creation is denied.
- The second level represents a *logical link*, grouping defined access devices together. On Clearspan, the link is modeled by a set of one or more access devices. Such sets can be created for enterprises and service provider groups. This level can be used to impose further restrictions to specific subscribers using specific access devices.

However, a service provider or enterprise can define a white list. When a member of a Session Admission Control group makes a call that matches the service provider/enterprise's white list, the call is allowed to complete even if the Session Admission Control group has reached its capacity.

CONFIGURATION

The system administrator can activate a subscriber-level SAC and assign a maximum number of sessions to an enterprise/service provider. The default maximum number of sessions must be provided on activation. Service provider/enterprise administrators can activate subscriber-level SACs and assign a maximum number of sessions to a group, equal or less than the maximum set for the enterprise/service provider. An option determines whether sessions that both originate and terminate within the same service provider/enterprise/group are counted. This option is added to the service provider/enterprise/group.

Enterprise and group (in service provider) administrators can create SAC groups, define access devices that belong to the groups, and impose call limits on these groups. In addition, an SAC group can be designated as the default SAC group for the organization. If an SAC group is designated as a default, new devices are automatically added to that SAC group on creation.

Service provider and enterprise administrators can define a white list, that is, a set of digit patterns to compare against outgoing calls.

SERVICE PACKS

This feature allows service providers to create groups 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 create packages of services that they can authorize and assign according to their marketing strategy. Service packs are managed by service providers and do not impact how system providers authorize services to service providers.

A service pack consists of a name, a description that is visible to group administrators, the list of user services, and the maximum number of packs that can be deployed to groups. Service packs consume the quantities of individual services assigned to the service provider by the system administrator as soon as they are created.

Service Pack Strategies

The following are examples of service pack creation strategies that can be used by service providers. All these strategies are allowed by Clearspan; however, they are not enforced.

- **Exclusive packs** – 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.
- **Comprehensive packs** – 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.
- **Combination packs** – 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 provides the ability to extend comprehensive user packs with “special” services.
- **Unrestricted packs** – A service provider creates new service packs that contain some of the same services that exist in the old service packs. For example, the service provider can create package B with services 2, 3, and 4 to replace the existing package A that contains services 1, 2, and 3. The service provider assigns

service pack B to users with service pack A and then removes pack A. The reason for this is to migrate from package A to package B without users' losing their configuration data. The user ends up with services 2, 3, and 4, with services 2 and 3 remaining in their original configuration. If service pack A was removed before service pack B was added, the user would lose configuration data for services 2 and 3.

This strategy is useful for upgrading users from an "old" strategy to a "new" strategy. However, it is **not** recommended in general because it is difficult to maintain.

Service Pack Migration

The Service Pack Migration feature is a powerful tool that can be used to assign (or remove) services and service packs to groups of users.

It allows the administrator to convert in bulk large groups of users that match 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.

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 also allows a service provider to deprecate old marketing schemes.

Service providers authorize and unauthorize a desired quantity of service packs to the group. A service provider can assign as many user packs as desired to a group, including an unlimited number. They can remove service packs unless service packs are assigned to a user.

Group administrators can assign and remove service packs to and from users. The services available to a user are all the services in the packs and individual services assigned to the user. When a pack is removed, the services in the pack are no longer available to the user and the user loses any configuration data associated with these services, unless the service is assigned individually or in another pack.

Group administrators cannot individually assign services included 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 without using service packs, as well as in addition to using service packs.

SHARED DEVICES

Clearspan allows shared identity/device profiles to be defined at the system, service provider/enterprise, and group levels.

DESCRIPTION

This feature allows for sharing access devices across organizations, which is especially useful when supporting large access devices. It can also be used to support configurations in which Clearspan is used to provide voice mail for another host system (in which case the “host” system is configured as a shared access device for the purpose of delivering the message waiting indicator notifications).

CONFIGURATION

Shared devices can be created at the system, service provider, and group levels making them available within the scope in which they are defined. The administrator can then assign these devices to users or services.

TRUNK BULK NUMBER PROVISIONING

Trunk Bulk Number Provisioning allows for bulk provisioning ranges of numbers (trunk users) against a trunk group. This feature provides an automated process to add users/lines to existing trunk groups. These lines can be created by using a selected list of phone numbers or ranges of extensions. In addition, the services or service packs to be assigned to these lines can be selected.

Using the web portal, the administrator specifies the details of the user creation task and launches the task. When the task is complete, a detailed report of the task is available.

The ability to add user creation tasks is available to administrators who have permission to add users, associate users to devices, assign phone numbers/extensions to users, and assign services to users. Viewing/deleting a user creation task is available to all administrators.

DEFINE USER CREATION TASK

Users can be created based on either directory numbers (DNs) or extensions. If DNs are used, the user ID format, line/port format, and contact format can be specified as either the extension, the national DN (no country code), or the E.164 version of the DN. The domain for the user ID and line/port can be selected from any that are assigned to the group. In this case, the extension can (optionally) be populated based on the DN and the extension length setting for the group. The contact can (optionally) be set if the trunk group is on a device that supports static registrations.

If extensions are used, the user ID, line/port, and contact may only be based on the extension. The contact can (optionally) be set if the trunk group is on a device that supports static registrations. The domain for the user ID and line/port can be selected from any that are assigned to the group.

The department, time zone, and language of the trunk group are used for the users created. The DN or extension is used for the Hiragana first name/last name, if Hiragana support is enabled. The password is either the DN or the extension and the password rules do not prevent the user from being added.

The users created are assigned the services and service packs selected on the page. The list of services and service packs available are determined from those that are authorized to the group. The users created do not receive the services and service packs that are part of the new user services template set for the group.

Once the user creation task has been created, it is viewable, but not modifiable until it has completed. After completion, the log file is available and the task can then be deleted.

RUN USER CREATION TASK

User creation tasks are run one at a time on a first-come first-serve basis. The status of a task is Pending, Executing, or Completed. As the task is running, each user is created along with the assignment of the desired services and service packs, and then there is a small delay of no more than half of a second.

Each user and their service assignment is part of a single logical transaction. If for any reason, there is an error during the adding of a single user (for example, due to duplicate user ID, DN already in use, or duplicate line/port) or assigning any of the services or service packs to a single user, that user fails. The user creation task continues to attempt to create each user until all of the DNs or extensions allocated have been exhausted. If the server is shut down or stops suddenly, then the user creation task is stopped and it picks up where it left off when the server restarts.

VERIFY TRANSLATION AND ROUTING

The Verify Translation and Routing (VTR) web tool allows administrators to run test calls to gather information about translation, routing, and services for calls.

DESCRIPTION

The Verify Translation and Routing tool provides a translation, routing, and service usage analysis that can be used without having to use a device to originate or to terminate a test call on behalf of or to a user.

The main purpose of the tool is to allow administrators to quickly determine the following:

- Which specific rule allowed the detection of the originating user
- Whether the translation results are for the Application Server only or whether they involve Network Server translations (this includes any kind of service triggered or network routing translations)
- Which originating services have executed
- Which terminating services have executed
- Why a call attempt was blocked, and then by which service or which policy
- What treatment was used, if applicable
- Whether a redirection occurred and to what destination

All acceptable combinations of Clearspan call originations and terminations, as well as, public-switched telephone network (PSTN) originations and terminations are supported.

The tool is available to authorized administrators from the web portal. Administrators can create VTR requests by entering parameters for the test call, such as origination and destination of the call, or by entering a SIP message to trigger the call. Test results are displayed on the same page.

Depending on the selected parameters, validation occurs to indicate errors before requests are sent. Validation does not prevent a submission of erroneous requests to allow administrators see what would happen if incorrect data was sent.

The system administrator can also perform test calls via the Application Server command line interface.

CONFIGURATION

To use the tool, administrators other than the system administrator must be authorized to do so via Administrator policies.

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