Cisco BroadWorks

Service Guide

Release 23.0

Document Version 2

Notification

The BroadSoft BroadWorks has been renamed to Cisco BroadWorks. Beginning in September 2018, you will begin to see the Cisco name and company logo, along with the new product name on the software, documentation, and packaging. During this transition process, you may see both BroadSoft and Cisco brands and former product names. These products meet the same high standards and quality that both BroadSoft and Cisco are known for in the industry.

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Document Revision History

Release	Version	Reason for Change	Date	Author
14.0	1	Updated document for rebranding.	March 8, 2006	Roberta Boyle
14.0	1	Updated document for EVs 28669 and 24711: Details on Hand Raising and Roll Call Conferencing Features.	March 15, 2006	Robb Surridge
14.0	1	Updated document for the following Release 14.0 features: ■ Call Forwarding No Answer user service (modified to mention new FACs) ■ Call Waiting user service (modified to include Distinctive Ringback and new FACs) ■ Calling Line ID Blocking user service (modified to mention new FACs) ■ Charge Number user service (added) ■ N-Way Calling user service (added) ■ Physical Location user service (added) ■ Privacy user service (added) ■ Sequential Ringing user service (modified to mention new FACs) ■ Service Scripts User service (added) ■ User Intercept user service (modified for Emergency Intercept feature) ■ Service Scripts Group service (added) ■ Fax Messaging service (added) ■ Third-Party Voice Messaging service (modified to mention new FACs) ■ Voice Messaging User service (modified to mention new FACs) ■ Voice Portal Messaging service (modified to mention new FACs)	March 28, 2006	Robb Surridge
14.0	1	Updated document for EV 30420: SCF, SCA, and SCR features modified with note that 10-digit numbers may be required.	April 11, 2006	Robb Surridge
14.0	1	Made minor correction for Hunt Group service interactions for Call Waiting.	May 2, 2006	Patricia Renaud
14.0	1	Deleted custom features.	May 22, 20006	Patricia Renaud
14.0	1	Edited changes and updated styles.	June 9, 2006	Patricia Renaud
14.0	1	Added paragraph to CFA service, describing possible loop scenario, Windows Messenger incompatibility with MCA service, widened scope of CLID Blocking Persistence service to enterprise, maximum call center Music On Hold length/file size. Updated description and formatting of Speed Dial 100 conflicts with existing FACs.	June 20, 2006 June 21, 2006	Robb Surridge
14.0	2	Added clarification to Third-Party Voice Mail Support to indicate that the service must be turned on for users independently of group settings (EV 34657).	August 30, 2006	Robb Surridge

Release	Version	Reason for Change	Date	Author
14.0	2	Added clarification to Incoming Calling Plan "Calls from outside the group" and "Calls from within the group" selection criteria (EV 34528).	August 31, 2006	Robb Surridge
14.0	2	Edited document. For Selective Call Acceptance and Selective Call Rejection services, updated each description to indicate that the service provides a treatment upon blocking a call.	October 6, 2006	Patricia Renaud
14.0	2	Updated document for the following Release 14.0 features: ▪ Directory Number Hunting service and Call Waiting Support for Agents (EVs 25770 and 29731) ▪ Call Center Support for Agent and Supervisor Clients and Call Center Reporting (EV 30848) ▪ Additional FAC support (EV 29251) ▪ SCA Bridging (EV 10859) ▪ Emergency Intercept (EV 27789) ▪ Trunk Bulk Number Provisioning (EV 27242) ▪ Data Validation at Provisioning (EV 10823) ▪ CMS Integration (EV 32431)	October 10, 2006	Patricia Renaud
14.0	2	Updated document for the Release 14.0 Two-Stage Dialing feature (EV 37064) and incorporated Product Management review comments.	November 8, 2006	Patricia Renaud
14.0	3	Updated service maximum recording lengths for EV 42348, in sections: *3.44 Custom Ringback User* *4.3 Auto Attendant* *4.10 Custom Ringback Group* *5.6.1.1 Message Deposit*.	March 29, 2007	Roberta Boyle
14.0	3	Edited changes.	April 4, 2007	Patricia Renaud
14.0	4	Added Mobile Extension-to-Extension Dialing service.	May 9, 2007	Robb Surridge
14.0	4	Consolidated descriptions for Cancel Call Waiting (Persistent and Per Call) with the Call Waiting description. As requested, removed description for the Mobile Extension-to-Extension Dialing service.	June 21, 2007	Patricia Renaud
14.0	4	Edited and published document.	June 21, 2007	Patricia Renaud
14.0	5	Added section *4.8.1.5 Interaction between ODP, OCP, and OPDP* for EV 46519.	July 3, 2007	Roberta Boyle
14.0	5	Edited changes and published document.	July 9, 2007	Andrea Fitzwilliam
14.0	6	Made a small change to the *Conferencing* section for EV 56669.	December 12, 2007	Roberta Boyle
14.0	6	Updated section *3.87 Two-Stage Dialing* for EV 57910.	January 15, 2008	Roberta Boyle

Release	Version	Reason for Change	Date	Author
14.0	6	Updated section *4.8.1.5 Interaction between ODP, OCP, and OPDP* for EV 56039.	January 24, 2008	Roberta Boyle
14.0	6	Updated section *3.10.2 Configuration* for EV 58517.	February 5, 2008	Roberta Boyle
14.0	6	Added section *3.87.2.1 Two-Stage Dialing Prompt* for EV 58760.	February 20, 2008	Roberta Boyle
14.0	6	Updated section *4.6 Call Park* for EV 59425.	February 20, 2008	Roberta Boyle
14.0	6	Edited changes and published document.	April 3, 2008	Andrea Fitzwilliam
15.0	1	Updated document for Release 15.0 (including content for Release 14.sp1 through Release 14.sp6).	May 29, 2008	Gillian Horgan
15.0	1	Edited document.	August 10, 2008	Patricia Renaud
16.0	1	Updated document for Release 16.0.	September 15, 2008	Gillian Horgan
16.0	1	Updated section *4.18.1.2 User Services* (Hunt Groups) for EV 66679. Updated section *3.86.1.2 IP Phone* for EV 64613.	October 30, 2008	Goska Auerbach
16.0	1	Updated sections *3.81 Shared Call Appearance (Multiple Appearance Directory Numbers)*, *3.66 N-Way Calling*, and *5.6.1.4 Message Retrieval* to incorporate changes requested in the following EVs: 66970, 68557, 59945, and 65785.	December 10, 2008	Goska Auerbach
16.0	1	Updated section *4.25 Resource Inventory Reporting* for EV 69617.	December 22, 2008	Roberta Boyle
16.0	1	Updated sections *3.44.1 Description* and *4.10 Custom Ringback Group* for EV 88142.	January 7, 2009	Roberta Boyle
16.0	1	Updated section *3.35 CommPilot Express* for EV 89983.	May 15, 2009	Roberta Boyle
16.0	1	Edited changes to document (for EVs).	June 2, 2009	Andrea Fitzwilliam
16.0	1	Edited changes to document.	July 20, 2009	Andrea Fitzwilliam
16.0	1	Updated document with additional Release 16.0 content.	August 19, 2009	Goska Auerbach
16.0	1	Reviewed changes to document.	August 21, 2009	Patricia Renaud
16.0	1	Edited and published document.	August 29, 2009	Patricia Renaud
17.0	1	Updated document for Release 17.0.	October 8, 2009	Goska Auerbach
17.0	1	Updated sections *3.63 Malicious Call Trace*, *3.72 Prepaid*, *4.18.1.4 Directory Number Hunting*, and *5.4 SMDI Message Desk* for EV 101265.	November 2, 2009	Goska Auerbach
17.0	1	For EV 103567: ▪ Deleted section *3.78 Web Call Logs*. ▪ Added section on *Basic Call Logs*. ▪ Added section on *Enhanced Call Logs*.	December 10, 2009	Roberta Boyle

Release	Version	Reason for Change	Date	Author
17.0	1	Updated section *3.81 Shared Call Appearance (Multiple Appearance Directory Numbers)* for EV 104983.	January 25, 2010	Goska Auerbach
17.0	1	Updated the Call Transfer service description and modified sections *3.31 Call Transfer, 3.47 Directed Call Pickup with Barge-in,* and *3.86 Three-Way Calling* for EV 105962.	February 1, 2010	Goska Auerbach
17.0	1	Updated document for Release 17.0 and for EV 110776.	April 28, 2010	Goska Auerbach
17.0	1	Updated section *5.7.1.5 System and Network Voice Portal* for EV 109343.	April 29, 2010	Eric Bernier
17.0	1	Edited and published document.	April 29, 2010	Margot Hovey-Ritter
17.0	2	Made minor editorial changes.	May 5, 2010	Patricia Renaud
17.0	2	Updated sections *4.2 Attendant Console* and *4.3 Auto Attendant* for EV 111016.	May 10, 2010	Goska Auerbach
17.0	2	Edited changes and published document.	June 9, 2010	Andrea Fitzwilliam
18.0	1	Updated document for Release 18.0.	June 21, 2010	Goska Auerbach
18.0	1	Updated section *4.18 Hunt Group* for EV 114538.	July 12, 2010	Goska Auerbach
18.0	1	Updated sections *3.11 Call Forwarding Always*, *3.14 Call Forwarding Busy*, *3.16 Call Forwarding No Answer*, and *3.18 Call Forwarding Not Reachable* for EV 115496.	January 31, 2011	Goska Auerbach
18.0	1	Updated section *3.80 Sequential Ringing* for EV 129767.	April 19, 2011	Goska Auerbach
18.0	1	Updated section *4.18 Hunt Group* for EV 127562.	July 8, 2011	Goska Auerbach
18.0	1	Updated section *4.8 Calling Plan* for EV 145040.	August 23, 2011	Goska Auerbach
18.0	1	Updated section *4.18.1.2 User Services* (Hunt Groups) for EV 132016.	October 17, 2011	Goska Auerbach
18.0	1	Updated section *3.11 Call Forwarding Always* for EV 149492. Updated section *4.8.1.1 Call Topology* for EV 140788. Updated section *3.59 Intercept User* to address EV 153754.	November 16, 2011	Goska Auerbach
18.0	1	Updated document with comments from review.	January 3, 2012	Goska Auerbach
18.0	1	Added sections *6.16 Schedules* and *6.17 Selective Criteria* for EV 153826.	January 12, 2012	Goska Auerbach
18.0	1	Edited and published document.	January 26, 2012	Andrea Fitzwilliam
18.0	2	Updated section *3.31 Call Transfer* for EV 157676 and section *4.17 Group Paging* for EV 157684.	March 16, 2012	Goska Auerbach

Release	Version	Reason for Change	Date	Author
18.0	2	Updated section *3.47 Directed Call Pickup with Barge-in* for EV 130226.	March 19, 2012	Joel Collin
18.0	2	Edited changes and published document.	April 2, 2012	Jessica Boyle
18.0	3	Added section *4.8.1.6 Interaction with Virtual Users* for EV 168552.	June 27, 2012	Joel Collin
18.0	3	Edited changes and published document.	July 3, 2012	Andrea Fitzwilliam
19.0	1	Updated document for Release 19.0.	July 18, 2012	Goska Auerbach
19.0	1	Updated section *4.6 Call Park* for EV 174839.	September 20, 2012	Goska Auerbach
19.0	1	Updated section *4.6 Call Park* for EV 175336.	October 2, 2012	Goska Auerbach
19.0	1	Made minor editorial changes to the document.	November 27, 2012	Patricia Renaud
19.0	1	Updated document with information about the availability of services and features in AS mode and XS mode.	December 4, 2012	Goska Auerbach
19.0	1	Edited changes and published document.	December 13, 2012	Patricia Renaud
20.0	1	Updated document for Release 20.0.	January 7, 2013	Goska Auerbach
20.0	1	Updated section *3.81 Shared Call Appearance (Multiple Appearance Directory Numbers)* for EV 182776.	February 21, 2013	Goska Auerbach
20.0	1	Updated section *4.1.1.2 Feature Access Code-Based Usage* for EV 197481.	October 29, 2013	Goska Auerbach
20.0	1	Edited changes to document.	November 27, 2013	Patricia Renaud
20.0	1	Removed 3G/4G Continuity service and information on services availability in XS mode, since XS mode is not part of Release 20.0.	December 11, 2013	Goska Auerbach
20.0	1	Edited changes and published document.	December 12, 2013	Patricia Renaud
21.0	1	Updated document for Release 21.0.	March 12, 2014	Goska Auerbach
21.0	1	Updated the BroadSoft legal notice.	October 15, 2014	Goska Auerbach
21.0	1	Edited changes and published document.	January 8, 2015	Andrea Fitzwilliam
22.0	1	Updated document for Release 22.0.	May 11, 2015	Goska Auerbach
22.0	1	Updated sections *3.16 Call Forwarding No Answer*, *3.35 CommPilot Express*, *3.80 Sequential Ringing*, *5.5 Third-Party Voice Mail Support*, *5.6 Voice Messaging*, and *6.3.1.3 Call Limits* for PR-47783.	August 27, 2015	Goska Auerbach
22.0	1	Updated section *5.6.1.1 Message Deposit*.	October 2, 2015	Karine Leduc
22.0	1	Updated section *3.47.1.1 Barge-in* for PR-48466.	January 15, 2016	Goska Auerbach
22.0	1	Updated sections *3.84 Speed Dial 8* and *3.85 Speed Dial 100* for PR-49578.	February 8, 2016	Goska Auerbach

Release	Version	Reason for Change	Date	Author
22.0	1	Clarified in the 3G/4G Continuity user service configuration that the service is enabled by default when assigned to a user.	March 21, 2016	Nathalie Houle
22.0	1	Updated section *3.19.2.2 Feature Access Codes* for PR-50988.	August 22, 2016	Goska Auerbach
22.0	1	Updated section *3.81.6 Call Retrieve* for PR-51607.	October 11, 2016	Goska Auerbach
22.0	1	Updated sections *3.9 BroadWorks Mobility* and *3.81 Shared Call Appearance (Multiple Appearance Directory Numbers)* for PR-50339.	October 31, 2016	Goska Auerbach
22.0	1	Edited changes.	December 5, 2016	Patricia Renaud
22.0	1	Reviewed and approved document.	December 6, 2016	Jim McGonigle
22.0	1	Edited and published document.	December 6, 2016	Patricia Renaud
22.0	2	Updated section *3.16 Call Forwarding No Answer* for PR-52909.	December 14, 2016	Goska Auerbach
22.0	2	Updated section *3.81.9 Configuration* to clarify that the configuration of the bridging and Call Park settings is unrelated to the Multiple Call Arrangement service for PR-54183.	January 17, 2017	Karine Leduc
23.0	1	Updated document for Release 23.0.	January 27, 2017	Goska Auerbach
23.0	1	Added section *2.2 Services Assignable to Virtual Users* for PR-48198.	April 18, 2018	Goska Auerbach
23.0	1	Restructured section *1 Summary of Changes*.	April 19, 2018	Goska Auerbach
23.0	1	Updated sections *3.19.2.2 Feature Access Codes* and *6.7.2 Configuration* for PR-57480.	April 25, 2018	Goska Auerbach
23.0	1	Updated section *3.54 Flexible Seating* for PR-58136.	May 22, 2018	Goska Auerbach
23.0	1	Updated sections *4.5.1.1 Call Center Offerings* and *4.5.1.2 Features* to specify that only agents with Call Center – Premium can be silently monitored for PR-58880.	June 8, 2018	Goska Auerbach
23.0	1	Updated section *3.26 Call Logs* for PR-58251.	November 14, 2018	Goska Auerbach
23.0	1	Edited changes, rebranded document, and published document.	November 30, 2018	Andrea Fitzwilliam
23.0	2	Updated document for PR-60400.	January 10, 2019	Goska Auerbach
23.0	2	Completed rebranding for Cisco.	March 12, 2019	Goska Auerbach
23.0	2	Edited changes and republished document.	March 20, 2019	Patricia Renaud

Abstract

This document provides a summary of user and group services offered by Cisco BroadWorks as of Release 23.0. For each service, this document provides an overview of the following:

- Features and functionality

- Usage and activation (as applicable)

- Parameters and configuration options (as applicable)

This document is intended for Cisco customers and partners and it is a complement to other Cisco BroadWorks product information.

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1   Summary of Changes

This version of the document includes the impact of new services or features on existing services.

1.1   Changes for Release 23.0

Changes for Release 23.0, Document Version 2

This version of the document includes the following changes:

- Added section *6.14 User No Answer Timer* and updated sections *3.16 Call Forwarding No Answer*, *3.17 Call Forwarding No Answer to Voice Mail*, *3.35 CommPilot Express*, *3.68 Personal Assistant*, *3.80 Sequential Ringing*, *5.5 Third-Party Voice Mail Support*, *5.6 Voice Messaging, and 6.3 Call Processing Policies* for PR-60400.

- Completed rebranding for Cisco.

Changes for Release 23.0, Document Version 1

This version of the document includes the following changes:

- Added and updated services:

    - GETS Calling Line ID Delivery Blocking

    - Terminating Alternate Trunk Identity

    - Resource Priority

    - Voice Portal

- Added section *2.2 Services Assignable to Virtual Users* for PR-48198.

- Updated sections *3.19.2.2 Feature Access Codes* and *6.7.2 Configuration* for PR-57480.

- Updated section *3.54 Flexible Seating* for PR-58136.

- Updated sections *4.5.1.1 Call Center Offerings* and *4.5.1.2 Features* to specify that only agents with Call Center – Premium can be silently monitored for PR-58880.

- Updated section *3.26 Call Logs* for PR-58251.

1.2   Changes for Release 22.0

Changes for Release 22.0, Document Version 2

This version of the document includes the following changes:

- Updated section *3.16 Call Forwarding No Answer* for PR-52909.

- Updated section *3.81.9 Configuration* to clarify that the configuration of the bridging and Call Park settings is unrelated to the Multiple Call Arrangement service, for PR-54183.

Changes for Release 22.0, Document Version 1

This version of the document includes the following changes:

- Added services and capabilities:

    - Direct Route

    - Call Forwarding Always Secondary

- – Silent Alerting
- – Alternate User IDs
- – UC-One Communicator
- – Conference Room
- Updated services and capabilities:
  - – 3G/4G Continuity
  - – Calling Name Retrieval
  - – Schedules
  - – Call Processing Policies
  - – Intercept User
  - – Intercept Group
  - – Shared Call Appearance (Multiple Appearance Directory Numbers)
  - – Immediate SIP Connectivity for New Site (Route List)
  - – Security Classification
  - – Voice Portal
  - – Configurable Default Feature Access Codes
  - – Communication Barring – Fixed
- Removed description and references to services that are no longer supported:
  - – CommPilot Call Manager
  - – Attendant Console
  - – BroadWorks Connector for Lotus Sametime
  - – Service Scripts User
  - – Service Scripts Group
  - – UC-Connect
  - – LDAP Integration
  - – Outlook Integration
- Updated sections *3.16 Call Forwarding No Answer*, *3.35 CommPilot Express*, *3.80 Sequential Ringing*, *5.5 Third-Party Voice Mail Support*, *5.6 Voice Messaging*, and *6.3.1.3 Call Limits* for PR-47783.
- Updated section *3.47.1.1 Barge-in* for PR-48466.
- Updated sections *3.84 Speed Dial 8* and *3.85 Speed Dial 100* for PR-49578.
- Updated section *3.19.2.2 Feature Access Codes* for PR-50988.
- Updated section *3.81.6 Call Retrieve* for PR-51607.
- Updated sections *3.9 BroadWorks Mobility* and *3.81 Shared Call Appearance (Multiple Appearance Directory Numbers)* for PR-50339.

1.3   Changes for Release 21.0

Changes for Release 21.0, Document Version 1

This version of the document includes the following changes:

- Added services and capabilities:
  - Personal Assistant
  - Number Portability Announcement
  - Immediate SIP Connectivity for New Site (Route List)
  - Collaborate
- Modified services:
  - Advice of Charge
  - Configurable Default Feature Access Codes
  - BroadWorks Mobility
  - Alternate Numbers
- Updated the list of services available in AS and XS mode.

1.4   Changes for Release 20.0

Changes for Release 20.0, Document Version 1

This version of the document includes the following changes:

- Added services:
  - Executive
  - VoiceXML
  - Security Classification
  - Flexible Seating
  - Verify Translation and Routing
- Modified services:
  - Hierarchical Communication Barring
  - Call Center
  - Call Recording
  - BroadWorks Mobility
  - Auto Attendant
  - Hunt Group
  - N-Way Calling
  - Configurable Default Feature Access Codes
- Updated section *3.81 Shared Call Appearance (Multiple Appearance Directory Numbers)* for EV 182776.
- Updated section *4.1.1.2 Feature Access Code-Based Usage* for EV 197481.

■ Removed 3G/4G Continuity service and information on service availability in XS mode, since XS mode is not part of Release 20.0.

1.5   Changes for Release 19.0

Changes for Release 19.0, Document Version 1

This version of the document includes the following changes:

■ Added services:

   – Find-me/Follow-me

   – Group Night Forwarding

   – 3G/4G Continuity

■ Modified services:

   – Auto Attendant

   – Configurable Default Feature Access Codes

   – Hunt Group

   – Session Admission Control

   – Voice Portal

   – Meet-Me Conferencing

   – Hoteling

■ Updated section *4.6 Call Park* for EV 174839.

■ Updated section *4.6 Call Park* for EV 175336.

■ Added information about services availability in AS mode and XS mode.

1.6   Changes for Release 18.0

Changes for Release 18.0, Document Version 3

This version of the document includes the following change:

■ Added section *4.8.1.6 Interaction with Virtual Users* for EV 168552.

Changes for Release 18.0, Document Version 2

This version of the document includes the following changes:

■ Updated section *3.31 Call Transfer* for EV 157676.

■ Updated section *4.17 Group Paging* for EV 157684.

■ Updated section *3.47 Directed Call Pickup with Barge-in* for EV 130226.

Changes for Release 18.0, Document Version 1

This version of the document includes the following changes:

■ Updated section *4.18 Hunt Group* for EVs 114538 and 127562.

■ Updated sections *3.11 Call Forwarding Always*, *3.14 Call Forwarding Busy*, *3.16 Call Forwarding No Answer*, and *3.18 Call Forwarding Not Reachable* for EV 115496.

■ Updated section *3.80 Sequential Ringing* for EV 129767.

■ Updated section *4.8 Calling Plan* for EV 145040.

■ Updated section *4.18 Hunt Group* for EV 132016.

- Updated section *3.63 Malicious Call Trace* for EV 140330.

- Updated section *3.11 Call Forwarding Always* for EV 149492.

- Updated section *4.8.1.1 Call Topology* for EV 140788.

- Updated section *3.59 Intercept User* to address EV 153754.

- Added sections *6.17 Schedules* and *6.18 Selective Criteria* for EV 153826.

- Removed references to Instant Conferencing and replaced them with references to Meet-Me Conferencing, where applicable.

- Updated and revised document for Release 18.0 and updated it with comments from review.

- Added new services:

  - Group Paging

  - Dialable Caller ID

  - Meet-Me Conferencing

  - Call Me Now

  - BroadWorks Mobility

  - Call Recording

  - BroadWorks Connector for Lotus Sametime

- Updated services:

  - Call Park

  - Shared Call Appearance

  - BroadWorks Anywhere

  - Fax Messaging

  - Speed Dial 100

  - Basic Call Logs

  - Enhanced Call Logs

  - Calling Plan

  - Call Waiting

  - Voice Messaging

  - Call Center

  - Hunt Group

  - Configurable Feature Access Codes

- Discontinued services:

  - Conferencing (replaced by Meet-Me Conferencing)

1.7   Changes for Release 17.0

Changes for Release 17.0, Document Version 2

This version of the document includes the following changes:

- Updated sections *4.2 Attendant Console* and *4.3 Auto Attendant* for EV 111016.

Changes for Release 17.0, Document Version 1

This version of the document includes the following changes:

- Updated services:
  - Intercept User
  - Selective Call Rejection
  - Calling Name Retrieval
  - Call Forwarding Not Reachable
  - Voice Messaging – Personal
  - Third-Party Voice Mail Retrieval
  - Malicious Call Trace
  - Call Center
  - Extension Dialing
  - Configurable Extension Dialing
  - Customer Originated Trace

- Added new services:
  - Communication Barring User-Control
  - Calling Name Delivery
  - Calling Number Delivery
  - Classmark
  - Explicit Call Transfer
  - Virtual On-Net Enterprise Extensions
  - Route Point
  - Location-Based Calling Restrictions
  - Pre-alerting Announcement
  - Legacy Automatic Callback
  - Session Admission Control
  - Office Communicator Tab

- Updated sections *3.63 Malicious Call Trace*, *3.72 Prepaid*, *4.18.1.4 Directory Number Hunting*, and *5.4 SMDI Message Desk* for EV 101265.

- Deleted section *3.78 Web Call Logs* and added sections *3.26.1.1 Basic Call Logs* and *3.26.1.2 Enhanced Call Logs* for EV 103567.

- Updated section *3.81 Shared Call Appearance (Multiple Appearance Directory Numbers)* for EV 104983.

- Updated the Call Transfer service description and modified sections *3.31 Call Transfer*, *3.47 Directed Call Pickup with Barge-in*, and *3.86 Three-Way Calling* for EV 105962.

- Added sections *3.70 Polycom Phone Services* and *3.91 Zone Calling Restrictions*, and updated section *3.44 Custom Ringback User* for EV 110776.

- Updated section *5.7.1.6 Residential Voice Portal* for EV 109343.

1.8   Changes for Release 16.0

Changes for Release 16.0, Document Version 1

This version of the document includes the following changes:

- Updated document for Release 16.0.

- Updated section *3.35.1 Description* (CommPilot Express) for EV 89983.

- Updated section *4.25 Resource Inventory Reporting* for EV 69617.

- Updated sections *3.44 Custom Ringback User* and *4.10 Custom Ringback Group* for EV 88142.

- Updated section *4.18 Hunt Group* for EV 66679.

- Updated section *3.86.1.2 IP Phone* for EV 64613.

- Updated sections *3.81 Shared Call Appearance (Multiple Appearance Directory Numbers)*, *3.66 N-Way Calling*, and *5.6.1.4 Message Retrieval* to incorporate changes requested in the following EVs: 66970, 68557, 59945, and 65785.

1.9   Changes for Release 15.0

Changes for Release 15.0, Document Version 4

This version of the document includes the following changes:

- Updated section *3.35 CommPilot Express* for EV 89983.

Changes for Release 15.0, Document Version 3

This version of the document includes the following changes:

- Updated sections *3.44 Custom Ringback User* and *4.10 Custom Ringback Group* for EV 88142.

- Updated section *4.25 Resource Inventory Reporting* for EV 69617.

- Updated section *3.66 N-Way Calling* for EV 68557.

- Updated section *3.81 Shared Call Appearance (Multiple Appearance Directory Numbers)* for EV 66970.

- Updated section *5.6.1.4 Message Retrieval* (Voice Messaging) for EV 59945.

Changes for Release 15.0, Document Version 2

This version of the document includes the following changes:

- Updated section *3.86.1.2 IP Phone* for EV 64613.

- Updated section *5.6.1.4 Message Retrieval* (Voice Messaging) for EV 65785.

- Updated section *4.18 Hunt Group* for EV 66679.

Changes for Release 15.0, Document Version 1

This version of the document includes the following changes:

- The following services have been added:

  – BroadWorks Anywhere

  – BroadWorks Anywhere Web Portal

  – Call Forwarding Not Reachable

  – Connected Line Identification Presentation

  – Connected Line Identification Restriction

  – In-Call Service Activation

  – MWI Delivery to Mobile Endpoints

- This version of the document also includes the impact of new services or features on existing services.

2 Introduction

This document provides a description of the services offered by Cisco BroadWorks.

The following services are covered:

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

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

  - **Virtual user services** – These services are authorized to a group and they make use of a virtual user who performs an action upon receiving a call (for example, an Auto Attendant). Typically, a virtual user service allows more than one service instance (virtual user) to be created for a group.

  - **Multiuser services** – These services are authorized to a group and they enable functionality that involves selected users in the group, for example, the Call Pickup service.

  - **Group services** – These services provide functionality that applies to all users in a group, for example, an Outgoing Calling Plan (OCP).

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

- **Provisioning and configuration tools** – These capabilities and tools are intended for service provider and enterprise administrators to help them manage the service provider/enterprise resources and services.

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

This document does not cover interactions between services unless they are part of the service itself. For information about service interactions, see the *Cisco BroadWorks Service Interaction Guide*.

Cisco BroadWorks configuration and provisioning tasks are hierarchical and although not explicitly mentioned for each occurrence, any configuration or provisioning action available to an administrative level is available to higher-level administrators.

2.1 Service Availability in AS and XS Mode

In Release 18.0, a new mode was introduced called the Execution Server (XS) mode. In Release 21.0 some services or features are available only in one mode. The following table lists the services described in this document and specifies their availability in each mode in Release 21.0. A check mark in the mode column next to a service means that the service is available in that mode. A check mark in both (*AS Mode* and *XS Mode*) columns without any additional comments means that the service is the same in both modes. Differences in the service functionality between the two modes are stated in the row for the service.

Service Name	AS Mode	XS Mode
User Services		
3G/4G Continuity		✓
Advice of Charge	✓	✓
Alternate Numbers (Multiple Numbers per User)	✓	✓
Anonymous Call Rejection	✓	✓
Authentication	✓	✓
Automatic Callback	✓	✓
Automatic Hold/Retrieve	✓	
BroadWorks Anywhere	✓	✓
BroadWorks Mobility	✓	
Busy Lamp Field	✓	✓
Call Forwarding Always	✓	✓
Call Forwarding Always Secondary	✓	
Call Forwarding Always to Voice Mail	✓	✓
Call Forwarding Busy	✓	✓
Call Forwarding Busy to Voice Mail	✓	✓
Call Forwarding No Answer	✓	✓
Call Forwarding No Answer to Voice Mail	✓	✓
Call Forwarding Not Reachable	✓	✓
Call Forwarding Selective	✓	✓
Calling Line ID Blocking Override	✓	✓
Calling Line ID Delivery	✓	✓
Calling Line ID Delivery Blocking	✓	✓

Service Name	AS Mode	XS Mode
Calling Name Delivery and Calling Number Delivery	✓	
Calling Name Retrieval	✓	✓
Calling Party Category	✓	
Call Logs	Basic Call Logs, Enhanced Call Logs	Enhanced Call Logs
Call Me Now	✓	
Call Notify	✓	✓
Call Recording	✓	
Call Return	✓	✓
Call Transfer	✓	✓
Call Waiting	✓	✓
Charge Number	✓	✓
Collaborate	✓	
CommPilot Express	✓	✓
CommPilot Personal Web Portal	✓	✓
Communication Barring User-Control	✓	✓
Configurable Calling Line ID	✓	✓
Conference Room	✓	
Connected Line Identification Presentation	✓	✓
Connected Line Identification Restriction	✓	✓
Consultation Hold	✓	✓
Customer Originated Trace (Call Trace)	✓	✓
Custom Ringback User	✓	✓
Dialable Caller ID	✓	
Directed Call Pickup	✓	✓
Directed Call Pickup with Barge-in	✓	✓
Direct Inward/Outward Dialing	✓	✓
Diversion Inhibitor	✓	✓
Do Not Disturb	✓	✓
Executive	✓	
Explicit Call Transfer	✓	
Flash Call Hold	✓	

Service Name	AS Mode	XS Mode
Flexible Seating	✓	
GETS Calling Line ID Delivery Blocking		✓
Group Night Forwarding	✓	✓
Hoteling	✓	✓
In-Call Service Activation	✓	
Intercept User	✓	✓
Last Number Redial	✓	✓
Legacy Automatic Callback	✓	
Location-Based Calling Restrictions	✓	
Malicious Call Trace	✓	✓
Music/Video On Hold User	✓	✓
Number Portability Announcement	✓	
N-Way Calling	✓	✓
Office Communicator Tab	✓	
Personal Assistant	✓	
Physical Location	✓	✓
Polycom Phone Services	✓	✓
Pre-alerting Announcement	✓	✓
Prepaid	✓	✓
Priority Alert	✓	✓
Privacy	✓	✓
Push To Talk	✓	
Remote Office	✓	
Security Classification	✓	
Selective Call Acceptance	✓	✓
Selective Call Rejection	✓	✓
Sequential Ringing	✓	✓
Shared Call Appearance (Multiple Appearance Directory Numbers)	✓	✓ + Simultaneous Primary and Alternate Location Calls
Silent Alerting	✓	
Simultaneous Ringing Personal	✓	✓
Speed Dial 8	✓	✓
Speed Dial 100	✓	✓

Service Name	AS Mode	XS Mode
Three-Way Calling	✓	✓
Two-Stage Dialing	✓	
UC-One Communicator	✓	
Video Add-On	✓	
Yahoo! Messenger Online Status	✓	
Zone Calling Restrictions	✓	
Group Services		
Account Codes	✓	✓
Authorization Codes	✓	✓
Auto Attendant	Auto Attendant – Basic, Auto Attendant – Standard	Auto Attendant (Release 18.0 functionality)
BroadWorks Anywhere Portal	✓	✓
Call Center	✓	
Call Park	✓	✓
Call Pickup	✓	✓
Calling Plan	✓	✓ Except Outgoing Pinhole Digit Plan
Configurable Feature Access Codes	✓	✓
Custom Ringback Group	✓	✓
Department Administrative Layer	✓	✓
Device Inventory	✓	✓
Emergency Zones	✓	
Extension Dialing	✓	✓
Find-me/Follow-me	✓	
Group Calling Line Identity	✓	✓
Group Paging	✓	
Hunt Group	✓	✓
Instant Group Call	✓	
Intercept Group	✓	✓
Loudspeaker Paging	✓	
Meet-Me Conferencing	✓	
Music/Video On Hold	✓	✓
Outside Access Code	✓	
Resource Inventory Reporting	✓	

Service Name	AS Mode	XS Mode
Route Point	✓	
Series Completion	✓	
Virtual On-Net Enterprise Extensions	✓	✓
VoiceXML	✓	✓
Messaging Services		
Fax Messaging	✓	✓
MWI Delivery to Mobile Endpoint	✓	✓
Direct Voice Mail Transfer	✓	
SMDI Message Desk	✓	
Third-Party Voice Mail Support	✓	✓
Voice Messaging	✓	✓
Voice Portal	✓ + System and Network Voice Portals	✓ + Extended Scope Voice Portal
Provisioning and Configuration Tools		
Alternate User IDs	✓	
Business Trunking	✓	
Call Processing Policies	✓	✓
Classmark	✓	
Communication Barring – Fixed	✓	✓
CommPilot Web Portals	✓	✓
Direct Route	✓	
Enterprise Trunking	✓	
Hierarchical Communication Barring	✓	✓
Immediate SIP Connectivity for New Site (Route List)	✓	
Large Enterprise Support	✓	✓
Network Classes of Service	✓	✓
Resource Priority	✓	
Restricted Administrative Access	✓	✓
Schedules	✓	✓
Selective Criteria	✓	✓
Session Admission Control	✓	

Service Name	AS Mode	XS Mode
Service Packs	✓	✓
Shared Devices	✓	✓
Terminating Alternate Trunk Identity	✓	
Trunk Bulk Number Provisioning	✓	
Verify Translation and Routing	✓	

2.2 Services Assignable to Virtual Users

Some Cisco BroadWorks services are configured by creating service instances (also called virtual users or virtual subscribers). Virtual users have many of the same attributes as a regular user, and they can be assigned some user services.

The following table lists the services that can be configured as virtual users and specifies the services that can be assigned to each of them.

Service	Virtual Subscriber												
	Auto Attendant	BroadWorks Anywhere Portal	Call Center	Collaborate Bridge	Find-me/Follow-me Group	Flexible Seating Host	Group Paging Group	Hunt Group	Instant Group Call	Meet-Me Conferencing Bridge	Route Point	Trunk Group	VoiceXML
Alternate Numbers	√	√	√	√	√			√	√	√	√		√
Anonymous Call Rejection	√	√	√	√	√			√	√	√	√		√
Authentication						√							
Basic Call Logs	√	√	√		√	√		√			√		√
Call Forwarding Always	√	√	√	√	√			√	√	√	√		√
Call Forwarding Always Secondary	√	√	√	√	√		√	√	√	√	√		√
Call Forwarding Busy	√	√	√	√	√			√	√	√	√		√
Call Forwarding Not Reachable					√								
Call Forwarding Selective	√	√	√	√	√			√	√	√	√		√
Calling Line ID Delivery Blocking	√	√	√	√	√	√	√	√	√	√	√		√
Calling Name Retrieval			√	√				√		√	√		
Calling Party Category				√						√	√		
Call Me Now	√	√	√		√			√	√		√		√
Call Notify	√	√	√	√	√		√	√	√	√	√		√
Call Recording	√		√									√	√

Service	Virtual Subscriber												
	Auto Attendant	BroadWorks Anywhere Portal	Call Center	Collaborate Bridge	Find-me/Follow-me Group	Flexible Seating Host	Group Paging Group	Hunt Group	Instant Group Call	Meet-Me Conferencing Bridge	Route Point	Trunk Group	VoiceXML
Call Transfer				√						√			√
Charge Number	√	√	√	√	√	√	√	√	√	√	√		√
Classmark				√						√			
Communication Barring User-Control	√	√	√	√	√		√	√	√	√	√		√
Connected Line Identification Restriction	√	√	√	√	√			√	√	√	√		√
Custom Ringback User	√	√	√		√		√	√	√				
Custom Ringback User – Video	√	√	√		√			√	√				
Directory Number Hunting			√					√					
Diversion Inhibitor	√	√	√	√	√			√	√	√	√		√
Do Not Disturb	√	√	√	√	√			√	√	√	√		√
Enhanced Call Logs	√	√	√		√	√		√			√		√
External Custom Ringback	√	√	√		√			√	√				
Fax Messaging	√	√	√	√				√	√	√	√		√
Group Night Forwarding	√	√	√	√	√		√	√	√	√	√		√
Pre-alerting Announcement	√	√	√	√	√		√	√	√	√	√		√
Priority Alert	√	√	√					√	√		√		
Privacy	√	√	√	√	√	√	√	√	√	√	√		√
Resource Priority						√						√	√
Selective Call Acceptance	√	√	√	√	√		√	√	√	√	√		√

Service	Virtual Subscriber												
	Auto Attendant	BroadWorks Anywhere Portal	Call Center	Collaborate Bridge	Find-me/Follow-me Group	Flexible Seating Host	Group Paging Group	Hunt Group	Instant Group Call	Meet-Me Conferencing Bridge	Route Point	Trunk Group	VoiceXML
Selective Call Rejection	√	√	√	√	√		√	√	√	√	√		√
SMDI Message Desk								√					
Third-Party Voice Mail Support	√	√	√	√	√			√	√	√	√		√
Virtual On-Net Enterprise Extensions			√	√						√	√		√
Visual Device Management						√							
Voice Messaging User	√	√	√	√	√			√	√	√	√		√
Voice Messaging User – Video	√	√	√	√	√			√	√	√	√		√
Zone Calling Restrictions	√	√	√	√	√			√	√	√	√		√

3 User Services

This section describes the services offered to Cisco BroadWorks users.

User services are typically assigned to specific users by the group administrator via the group web portal, and are managed and configured by the user via the personal web portal. This information is not explicitly mentioned in each section; however, exceptions to this model are specified.

3.1   3G/4G Continuity

This service allows devices on an IP Multimedia Subsystem (IMS) network to connect to Cisco BroadWorks using SIP for call control and using circuit-switched (CS) access for media when located in (or moving to) a 2G/3G-only area.

3.1.1   Description

The 3rd Generation Partnership Project (3GPP) has specifications for IMS centralized services (ICS) and service continuity. This feature implements a subset of these specifications for scenarios where the user device enhanced for IMS centralized services is using the Gm reference point.

Centralization refers to scenarios where a call starts with the user already in a 2G/3G area, whereas continuity refers to scenarios where the user equipment (UE) starts in an LTE/4G area, with the call control and media both on the IP packet-switched (PS) network, but moves out of the Long Term Evolution (LTE)/4G coverage into a 2G/3G area.

End-user devices (smartphones) have evolved to include the ability to use SIP, and they can now exploit this protocol for call control using the LTE/4G network. However, a problem arises when these devices move in or out of an area that does not yet have LTE/4G coverage and only has 2G/3G access networks.

The 3G/4G Continuity service allows devices on an IMS network to connect to Cisco BroadWorks using SIP for call control and using CS access for media when located in (or moving to) a 2G/3G-only area.

This service also allows handovers from the circuit-switched to the packet-switched network for devices supporting dual radio voice call continuity (DRVCC).

3.1.2   Configuration

The service is enabled by default when assigned to a user.

The user must be assigned a device that supports service centralization and continuity.

3.2 Advice of Charge

The Advice of Charge (AoC) service allows a user to have charging information sent to their phone. The service only applies to call originations (both internal and external). The advice of charge information is sent in different ways depending on how the service is configured.

3.2.1 Description

When the user makes a call, the charge is calculated based on the tariff applicable at the time of the call. Depending on the service configuration, the charge information is sent to the user's device periodically during the call (AoC-D) or only at the end of the call (AoC-E).

The AoC-D information is a cumulative charge incurred from the start of the call until the moment the charging information is sent to the originating device. The delay between each advice of charge message is configurable at the system level. When the call is released, the final advice of charge message is sent. For AoC-E calls, the charge information is sent to the device only after the call is released.

The charge information is provided using currency or units depending on the applicable tariff. If tariff information is not available for a call leg, advice of charge information is not sent during the call. When the call is released, a "not available" message is sent to the originating device.

If the service is assigned but not activated, the user can activate it for the next call by dialing *34 (default) before making the call.

3.2.2 Configuration

The user has no access to service configuration.

The group administrator activates the service for the user and specifies whether charging information is sent to the user device periodically during call or only after the call.

The system administrator can configure the following system-wide parameters:

- The delay between messages sent to the originating device, that is, the time in seconds between AoC notifications during an AoC-D call.

- The handling of incoming AoC information, which determines whether AoC information received from the network, is ignored or processed and sent to the originating device.

- The source of cost information, which specifies where the Service Price Enquiry requests should be sent (if required). If the setting is not configured, inquiries are not sent.

- Whether the device is able to decode and process AoC information, configured per identity/device type. If the device is not AoC-capable, the AoC information is not sent to the device. To avoid interoperability issues, this option should only be set for devices that can process AoC information.

The service provider/enterprise and group administrators can override the following setting for their organization:

- The delay between messages sent to the originating device, that is, the time in seconds between AoC notifications during an AoC-D call.

3.3 Alternate Numbers (Multiple Numbers per User)

The Alternate Numbers service allows a user to have up to ten alternate phone numbers in addition to the main phone number and to assign one of four distinctive ring patterns to each alternate number. In addition, a distinctive call waiting tone is associated with each distinctive ring pattern. Another name for this service is Distinctive Ring.

3.3.1 Description

This service allows a user to have up to ten alternate phone numbers or extensions in addition to the main phone number. A user can be reached through any of the phone numbers. Each alternate phone number can be either a regular direct inward dialing number (DID) or an extension.

The user can be provisioned with alternate numbers without the main number, in which case the user cannot make calls.

The user can associate one of the four ring patterns with each alternate number: *Normal*, *Long-Long*, *Short-Short-Long*, and *Short-Long-Short*. A call to an alternate number results in the ring pattern configured for that number while a call to the main number results in the normal ring pattern. However, the support for a distinctive ring pattern depends on the capability of a user's device. If the device does not support a distinctive ring pattern, then the normal ring pattern is applied to all calls.

Each ring pattern has a distinctive call waiting tone associated with it. If a user is busy and has Call Waiting enabled, an incoming call to the main number results in the normal call waiting tone, while an incoming call to an alternate number results in the call waiting tone for that number.

3.3.2 Configuration

The group administrator configures alternate phone numbers for the user.

The user can select the ring pattern and the description for each alternate phone number.

3.4 Anonymous Call Rejection

The Anonymous Call Rejection service enables a user to reject calls from parties who have explicitly restricted their identities. Another name for this service is Anonymous Caller Rejection.

3.4.1 Description

This service enables users to reject calls from callers outside the user's group who have blocked their identity (phone number and name) with a calling identity delivery blocking service.

Callers with blocked identities are connected to an announcement and informed that the user does not accept calls with the caller's identity blocked. The user's phone does not ring and the user does not see or hear any indication of the attempted call.

This service does not apply to calls from within a group.

3.4.2 Configuration

The user can activate the service (block anonymous calls) or deactivate it (allow anonymous calls).

3.5 Authentication

The Authentication service provides authentication of sessions for Session Initiation Protocol (SIP)-based customer premises equipment to prevent unauthorized access to the system.

3.5.1 Description

This service performs authentication upon the registration of an IP phone. This ensures that the user of the device is authorized to gain access to the system. Call originations from unregistered phones are denied.

Each SIP registration request received by Cisco BroadWorks is challenged. Cisco BroadWorks uses the information received from the phone and its own copy of the user's credentials to authenticate the request. If the authentication succeeds, the registration is accepted. If the authentication fails, another authentication challenge is issued.

In addition, call originations can also be authenticated at predefined intervals.

Standard MD5 digest authentication is used and the authentication information is configured both in the phone and on Cisco BroadWorks.

3.5.2 Configuration

The service provider/enterprise administrator can allow or deny access to authentication information on a per-group basis. If access to authentication information is allowed, the user configures a user ID and password; otherwise, the administrator configures the service for the user.

The system administrator can configure the system to challenge only every n^{th} origination request (INVITE messages) received by the system.

3.6 Automatic Callback

The Automatic Callback (ACB) service allows a user to be notified when the previously called busy party becomes idle and is connected without having to redial the phone number.

Depending on system configuration, the Automatic Callback service can only be activated against Cisco BroadWorks users or against non-Cisco BroadWorks users as well.

3.6.1 Description

When a user calls another party and the call cannot be connected because the line is busy, the user is prompted to make an Automatic Callback request (for example, "The line you are calling is busy. Press **1** if you would like to be notified when the line becomes available"). On request from the user, the system monitors a busy party and automatically establishes a call for the user when the busy party becomes available.

To request Automatic Callback, the user enters the digit prompted for and then goes on-hook. The user is notified with special ringing when the previously called party becomes idle. If the user answers the call, call setup is automatically initiated toward the other party and the user is not required to redial the phone number.

The new call attempt is treated as an originating call attempt; it can receive busy treatment or it can be redirected. Both parties must be available for the system to attempt call setup.

The destination is considered busy or unavailable if it cannot receive a call. This means that if the call is redirected and the new location is busy, the caller is not prompted to activate Automatic Callback. Similarly, Automatic Callback cannot be requested if the call is rejected by Selective Call Rejection or Selective Call Acceptance.

A user can deactivate all outstanding Automatic Callback requests by entering the Automatic Callback deactivation feature access code.

3.6.2 Configuration

The user can activate or deactivate the service. The system administrator can configure the following system-wide parameters:

- The method used to check whether a busy party has become idle. The possible options are:
  - *notify only*
  - *notify if possible and use polling otherwise*

  The first option only allows callback requests against Cisco BroadWorks users. The other option allows callback requests against any user whose network location is known.
- The frequency with which a busy party is pooled when the pooling method is used to monitor a busy line
- The length of time that the busy line is monitored before the request is canceled
- The amount of time to wait before retrying a busy Automatic Callback originator (the owner of the Automatic Callback session)
- The maximum number of rings used when alerting the originator that the destination is available
- The maximum number of concurrent originating callback sessions per user and the maximum number of concurrent terminating callback requests against a user
- The activation digit, that is, the digit the user enters to make an Automatic Callback request

3.7 Automatic Hold/Retrieve

The Automatic Hold/Retrieve (AHR) service enables a user to automatically hold and retrieve incoming calls without using feature access codes. This service also allows the holding of calls without having to use a flash key, which is often not available on SIP devices.

It is especially useful for attendants managing a large volume of incoming calls by enabling them to hold calls by simply transferring them to dedicated parking stations.

This service is typically assigned to dedicated parking stations, that is, to user accounts reserved for holding or retrieving calls.

3.7.1 Description

This service is commonly used by receptionists operating attendant consoles. When a call terminates on the attendant console, the receptionist answers the call, gathers information from the caller, and then transfers the call to a dedicated station with the Automatic Hold/Retrieve service enabled. The call is automatically put on hold and the caller is played Music On Hold while waiting.

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 Per-Call Deactivation, or Call Pickup) are blocked. Emergency calls made from a device with the Automatic Hold/Retrieve service activated receive a busy signal.

3.7.2 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.

3.8 BroadWorks Anywhere

The BroadWorks Anywhere service allows a user to use one or more network locations, also referred to as BroadWorks Anywhere locations, as extensions to the user's profile.

3.8.1 Description

The user can seamlessly consolidate devices, including fixed and mobile devices, from different networks under a single Cisco BroadWorks account and operate a single set of services. When the user originates a call from a BroadWorks Anywhere location, the user's originating services are executed as if the call had been dialed from one of the user's primary locations.

This service makes the provisioning model "network-independent". Instead of provisioning a network phone number as a shared call appearance, the user configures the phone number as a BroadWorks Anywhere location (that is, a phone number in E.164 format).

Each BroadWorks Anywhere location can be associated with selective criteria, which determine when the location should be alerted for incoming calls.

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. If required, multiple criteria entries can be defined for each location. For more information, see section *6.18 Selective Criteria*.

When the incoming call does not satisfy the criteria defined for a location, the location is not alerted. In addition, when no criteria entries are defined for a location or all criteria entries are inactive, the location is not alerted.

The BroadWorks Anywhere user service works in conjunction with the BroadWorks Anywhere portal. For more information, see section *4.4 BroadWorks Anywhere Portal*.

3.8.2 Configuration

The user can specify one or more phone numbers as BroadWorks Anywhere locations. The user is not expected to add a BroadWorks Anywhere location in E.164 format. When a BroadWorks Anywhere location is not entered in E.164 format, it is normalized to E.164 format and then saved.

A feature access code is also available to replace the "+" sign while dialing the E.164 number.

BroadWorks Anywhere locations can be enabled or disabled from the web interface or by using the Location Control feature access codes.

To enable a location, the user dials *12 (default) from the location to enable; to disable a location, the user dials *13 (default) from that location.

Note that feature access codes for location control are shared between BroadWorks Anywhere and Shared Call Appearance.

The user can configure the following for each BroadWorks Anywhere location:

- Outbound alternate phone number (or SIP-URI)

- Whether to enable diversion inhibitor

- Whether answer confirmation is required

- Whether to use Cisco BroadWorks-based call control services

- Zero or more selective criteria entries used to determine whether a location should be alerted. Each entry can be individually activated or deactivated.

In addition to these options, the user can configure BroadWorks Anywhere locations to be alerted for Click To Dial calls and for Group Paging calls. These settings apply to all BroadWorks Anywhere locations.

The BroadWorks Anywhere service cannot be assigned to virtual users.

3.9 BroadWorks Mobility

The BroadWorks Mobility service integrates the mobile network with the business services environment of the enterprise. This provides a seamless experience for the end user, allowing them to have a consistent business service on their mobile phone and desk phones. It also offers the flexibility of using multiple mobile devices and deciding when and how to use them for business and personal calls.

3.9.1 Description

This service enables the user to use Cisco BroadWorks enhanced services from a mobile device. The service can anchor mobile calls to Cisco BroadWorks and fork terminating calls to a mobile handset, making the business mobile and desk look like a single entity in which callers are able to reach a BroadWorks Mobility user by dialing a single number (main or mobile).

The user can have multiple mobile devices and decide for each device whether calls should be anchored on Cisco BroadWorks, either permanently or on a per-call basis. For example, the user can permanently anchor their business mobile device on Cisco BroadWorks and deactivate it temporarily to make a personal call.

The user can decide which devices to alert for calls to their mobile devices, desk phone, and alternate locations and whether to use the mobile number or the fixed calling line ID for calls made from a mobile device.

The BroadWorks Mobility feature also runs as an overlay to the Shared Call Appearance service. Shared Call Appearance alerts the mobile whenever there is a mobile termination and the BroadWorks Mobility service is set to ring the mobile phone. The user's alternate locations are alerted in parallel to the user's mobile devices. However, if the user is already on an active call from/to the primary device, then incoming calls are only presented to the primary device, independent of the alerting policy set for the mobile.

In addition, users have the ability to use different mobile access networks.

BroadWorks Mobility offers the following solutions:

■ Mobile Enterprise – This is a combined fixed Centrex (hosted Private Branch Exchange [PBX]) and Mobile Centrex application delivering the Centrex feature set to the mobile and fixed devices.

■ Mobile Office – This is a pure Mobile Centrex (also known under the name Mobile PBX) application, which delivers a complete Centrex feature set to the 3G, circuit-switched mobile device.

3.9.1.1 Additonal Features

■ Mid-Call Features – The BroadWorks Mobility service can be used in conjunction with the In-Call Service Activation service to provide mid-call features to the mobile device. These features are Flash and Transfer.

■ Enterprise Dialing – The BroadWorks Mobility service allows users to call by extension within the enterprise regardless of the access device (mobile or desk). Thus, mobile-to-mobile extension calling between users who belong to the same enterprise is possible.

■ Voice Messaging – A user can have a single voice mailbox. For this purpose, it is recommended, but not mandatory, to disable the carrier's voice messaging so that the calls can be sent to the Cisco BroadWorks Voice Mail system when appropriate and can notify the mobile handset of voice mail messages.

■ Mobile PBX Users – For mobile PBX users who do not have a fixed phone number, a mobile number can be selected to use for calling line ID presentation when the call policy is set to use the user's fixed phone number as calling line ID.

3.9.1.2 Feature Access Codes

The user can activate or deactivate call anchoring and the use of the mobile phone number as the calling line ID through feature access codes dialed from the user's mobile device.

■ To anchor on Cisco BroadWorks all calls made from a mobile device, the user dials #23 (default) from that device. The system then plays a confirmation announcement and the user hangs up.

■ To deactivate call anchoring, the user dials #24 (default). The system then plays a confirmation announcement and the user hangs up.

■ To anchor a call on Cisco BroadWorks, the user dials *23 (default) from their mobile 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 use Cisco BroadWorks services as if they called from their desk phone.

■ To deactivate call anchoring for one call, the user dials *24 (default) from the mobile device before making a call. The system plays a confirmation tone followed by a dial tone. The user can then make the call as usual and the call is processed in the mobile network.

■ To use the mobile phone number the as calling line ID when calling from a mobile phone, the user dials #28 (default) from that mobile device. The system then plays a confirmation announcement and the user hangs up.

■ To use the fixed calling line ID when calling from a mobile phone, the user dials #29 (default) from that mobile device. The system then plays a confirmation announcement and the user hangs up.

■ To use the mobile phone number as the calling line ID for one call, the user dials *28 (default) followed by a destination number. If the user dials *28 without any destination number, the system prompts the user and collects the destination digits.

■ To use the fixed calling line ID for one call, the user dials *29 (default) followed by a destination number. If the user dials *29 without any destination number, the system prompts the user and collects the destination digits.

3.9.2 Configuration

The following configuration is performed at the system level and can be overridden at the service provider and group levels:

■ Enabling or disabling location services

■ Enabling or disabling mobile station roaming number (MSRN) lookup

■ Enabling or disabling mobile state checking

■ Allowing or denying call originations and/or terminations

■ Enabling or disabling announcement suppression

■ Enabling or disabling delivery of the originator's internal calling line ID (CLID) to the mobile device being alerted for intra-enterprise and intragroup calls

In addition, the system administrator configures IP multimedia routing number (IMRN)-related settings, service access codes, signaling IP addresses, and assigns a pool of Mobile Subscriber Directory Numbers to service providers/enterprises that can be assigned to groups and then to users as their mobile phone numbers.

The following configuration is performed for the user:

- Enabling or disabling the service (administrator only)

- Assigning mobile phone numbers (administrator only)

- Deciding whether the user can call their own mobile device (administrator only)

- Specifying whether the user can use call anchoring controls (administrator only)

- Specifying the phones to ring independently for each mobile device and for the desk phone

- For each mobile device:

  - Allowing or denying call originations and/or call terminations

  - Enabling the mobile for Click To Dial and/or paging calls

  - Enabling the mobile for calls from virtual users, such as call center

  - Enabling the use of the mobile number as calling line identity

  - Enabling Diversion Inhibitor

  - Specifying whether answer confirmation is required

  - Specifying whether Cisco BroadWorks-based Call Control services should be used

To use mid-call features, the user needs to be assigned the In-Call Service Activation service.

3.10 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.

3.10.1 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.

3.10.2 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.

3.11 Call Forwarding Always

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

3.11.1 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.

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

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

- The system detects that the redirection destination is bound to a known Cisco BroadWorks user who is already involved in the current communication.

- The system 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.

3.11.2 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 Cisco BroadWorks Voice Mail system.

3.11.2.1 CommPilot 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.

3.11.2.2 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 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*.

3.11.2.3 SIP Phone

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

3.11.2.4 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 CommPilot 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 Forwarding Always service. After configuring this service, the user may hang up, or continue to navigate through other options of the CommPilot voice portal.

3.12 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.

3.12.1 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.

3.12.2 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.

3.13 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.

3.13.1 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 and Call Forwarding Always Secondary services have precedence over Call Forwarding Always to Voice Mail service.

3.13.2 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 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.

3.14 Call Forwarding Busy

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

3.14.1 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.

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

3.14.2 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 Cisco BroadWorks Voice Mail system.

3.14.2.1 CommPilot 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.

3.14.2.2 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 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*.

3.14.2.3 SIP Phone

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

3.15 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.

3.15.1 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.

3.15.2 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 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.

3.16 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.

3.16.1 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 time-out 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 triggers 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.

Cusco BroadWorks 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 the system is not aware that the called party is off-hook.

3.16.2 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 Cisco BroadWorks Voice Mail system.

3.16.2.1 CommPilot 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.

The user can also configure the number of rings before the call is forwarded. Note, however, that this setting applies to all services with no-answer handling. For more information and for the list of services that share this setting, see section *6.14 User No Answer Timer*.

3.16.2.2 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 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. For the list of service that share this setting, see section *6.14 User No Answer Timer*.

3.16.2.3 SIP Phone

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

3.17 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.

3.17.1 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.

3.17.2 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* or Third-Party Voice Mail Support 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 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.

   This service does not set the number of rings before forwarding the call but shares it with all other no-answer handling services. For more information, see section *6.14 User No Answer Timer*.

3.18 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.

3.18.1 Description

This service forwards incoming calls to a specified destination when the user's device is not accessible by Cisco BroadWorks. 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.

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

3.18.2 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.

3.18.2.1 CommPilot 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.

3.18.2.2 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 status and destination of Call Forwarding Not Reachable, the user dials *63* (default).

3.18.2.3 SIP Phone

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

3.19 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.

3.19.1 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 section *6.18 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.

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

3.19.2 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.

3.19.2.1 CommPilot 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.

3.19.2.2 Feature Access Codes

The user can activate or deactivate 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.

3.20 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".

3.20.1 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 Cisco BroadWorks 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 the system, 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

3.20.2 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.

3.21 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 via the device (if capable).

3.21.1 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, then 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.

3.21.2 Configuration

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

3.22 Calling Line ID Delivery Blocking

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

3.22.1 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.

3.22.1.1 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.

3.22.1.2 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 *3.22.1.1 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.

3.22.2 Configuration

Users can activate or deactivate the Calling Line ID Delivery Blocking service via the CommPilot 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 status of Calling Line ID Delivery Blocking, the user dials *54* (default).

3.23 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.

3.23.1 Description

Calling Name Delivery and Calling Number Delivery are separate services, which allow delivery of calling name and number to Cisco BroadWorks 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 Cisco BroadWorks 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.

3.23.2 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.

3.24 Calling Name Retrieval

The Calling Name Retrieval service allows Cisco BroadWorks 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.

3.24.1 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, the system 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 Cisco BroadWorks 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 Cisco BroadWorks user known in the Calling Name Retrieval subscriber's Application Server.

3.24.2 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 time-out

- 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.

3.25 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.

3.25.1 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 Cisco BroadWorks user with the same service in a different group.

Depending on the trunking device the Cisco BroadWorks system is interacting with for a given call, the system 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 Cisco BroadWorks.

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.

3.25.2 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 Cisco BroadWorks system.

3.26 Call Logs

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

The Basic Call Logs service provides a user with access to a small number of call logs while the Enhanced Call Logs service provides a user with access to their call history, allowing for storing a much larger number of call logs.

3.26.1 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).

3.26.1.1 Basic Call Logs

For 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.

Users who need access to a larger number of call logs should be assigned the Enhanced Call Logs service.

The following information is provided to the user:

- Placed Call Logs

  - *Name* – Shows the name of the called party, if available. It may also contain "Unavailable" or "Private".

  - *Phone Number* – Shows the called number. This field may contain a SIP-URI.

  - *Date/Time* – Shows the date and time of the call.

- Received/Missed Call Logs

  - *Name* – Shows the calling line identity name, if available. It may also contain "Unavailable" or "Private".

  - *Phone Number* – Shows the calling line identity number. It may also contain a SIP-URI, "Unavailable", or "Private".

  - *Date/Time* – Shows the date and time of the call.

3.26.1.2 Enhanced Call Logs

The following is a high-level overview of the key capabilities of the Enhanced Call Logs service. For information, see the *Enhanced Call Logs Solution Guide.*

The Enhanced Call Logs service allows for storing call logs for a much larger period of time than Basic Call Logs. Call logs are maintained on the Enhanced Call Logs Repository for a specific period of time called the retention period, which can be from 1 through 12 months.

The Enhanced Call Logs Repository stores relevant call log information; however, not the complete call detail record (CDR) information. Pertinent call log information, such as call

redirection by a Cisco BroadWorks user, is captured in the call log entries on the Enhanced Call Logs Repository.

The Enhanced Call Logs service provides more information about the call than the Basic Call Logs service, for example, call duration or the phone number to which the call has been redirected, if applicable. For information, see the *Enhanced Call Logs Solution Guide.*

Call logs are deleted in one of two ways:

■ A user has the ability to delete specific call logs. The deleted call logs cannot be retrieved from the Enhanced Call Logs Repository.

■ All call logs older than the configured retention period are deleted from the Enhanced Call Logs Repository.

3.26.2 Configuration

The Basic Call Logs service has no configuration.

For information on the configuration of the Enhanced Call Logs service, see the *Enhanced Call Logs Solution Guide*.

3.27 Call Me Now

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

3.27.1 Description

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

Cisco BroadWorks then screens the provided number using the selective criteria set 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 section *6.18 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, the system initiates a call to the provided number. From the external party's perspective, the call is a regular incoming call from the Cisco BroadWorks user.

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

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

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

Note that since the Cisco BroadWorks 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.

3.27.2 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.

3.28 Call Notify

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

3.28.1 Description

The user can define selective criteria that cause certain incoming calls to trigger an email 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 section *6.18 Selective Criteria*.

If the screening is successful, an email notification is sent to the specified address.

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

3.28.2 Configuration

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

3.29 Call Recording

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

3.29.1 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 Cisco BroadWorks 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 Cisco BroadWorks 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".

3.29.1.1 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.

3.29.1.2 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 Cisco BroadWorks 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.

3.29.1.3 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.

3.29.1.4 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.

3.29.1.5 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.

3.29.2 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.

3.30 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.

3.30.1 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 Cisco BroadWorks. 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.

3.30.1.1 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 configurable confirmation digit. The user can dial the digit while the announcement is being played.

An incorrect confirmation digit causes the announcement to be repeated. When an incorrect digit is entered twice or no digit is provided within seven seconds, the Call Return service terminates the call after instructing the user to hang up.

In the two-level mode, a request to return a call to a restricted number is processed as follows:

- If the user has a Calling Line ID Delivery service and if returning a call to a restricted number is allowed, the restricted number announcement is played and the user is instructed to dial a confirmation digit before the call to the restricted number is originated. If returning a call to a restricted number is not allowed, the call is denied and the restricted number denial announcement is played.

- If the user does not have a Calling Line ID Delivery service, the restricted number announcement is played and the user is instructed to dial a confirmation digit to confirm the request whether returning a call to a restricted number is allowed.

3.30.2 Configuration

The system administrator configures the following system-wide settings via the command line interface (CLI):

- Whether Call Return applies only to unanswered calls or to both answered and unanswered calls.

- Whether Call Return is allowed to a destination answered with Call Return.

- Whether returning a call to a restricted number is allowed.

- Whether two-level mode is allowed.

- When the two-level mode is enabled, further customization is possible:

    - The service can be configured to announce the date and time when the last incoming call was received.

    - A confirmation digit can be configured.

3.31 Call Transfer

The Call Transfer service enables the user to transfer a call to a specified destination. Call transfers can be blind, with third-party consultation, or with three-way consultation. In addition, the Busy Camp On and Call Transfer Recall features allow the user to camp the call to transfer on a busy destination and to be recalled if the transferred call is not answered for any reason. Another option, Diversion Inhibitor, enables the user to prevent transferred calls from being redirected.

3.31.1 Description

The user can transfer a call from a device after answering the call.

3.31.1.1 Blind Call Transfer

In this scenario, the user transfers an active call to a specific destination without consulting the destination party.

To initiate a Blind Call Transfer, the user with an active answered call presses the flash hook and calls the destination number. While the call is ringing, the user hangs up. The original caller hears ringing and is connected to the destination party, if the party answers the phone.

3.31.1.2 Transfer with Third-Party Consultation

In this scenario, the user consults with the add-on party before transferring the call to the add-on party.

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

The user can elect to abort the transfer during the consultation by flashing the switch hook twice, which releases the add-on party.

3.31.1.3 Transfer with Three-Way Consultation

Call Transfer with Three-Way Consultation allows a user to make a three-way call with the original caller and an add-on party before transferring the caller to the add-on party.

To initiate Call Transfer with Three-Way Consultation, the user presses the flash hook, receives a dial tone, and then dials the add-on party. When the call is answered, the user presses the flash hook and forms a three-way call with the add-on party and the original caller. To transfer, the user hangs up; this then transfers the original caller to the add-on party.

The user can elect to abort the transfer during the three-way consultation by flashing the switch hook, which releases the add-on party.

3.31.1.4 Blind Call Transfer versus Transfer with Consultation

The system performs a different type of validation for a blind call transfer than for a transfer with consultation.

Let us assume that User A calls User B, and User B then wants to transfer the call to User C. If user B initiates a blind transfer, the system only validates whether User B can transfer the call from User A to User C.

However, if user B tries to transfer the call with consultation, the system verifies whether user B can transfer calls from User A to User C **and** from User C to User A. The call will proceed only if transfer in both directions is allowed.

3.31.1.5 Call Transfer Recall

The Call Transfer Recall feature allows a transferred call to be reconnected to the transferring party if it reaches a failure or no-answer condition after transfer. Call Transfer Recall applies to both blind transfers and transfers with consultation. It does not apply to a transfer with consultation of a barge-in conference.

3.31.1.6 Busy Camp On

The Busy Camp On feature allows a user to camp a call against a busy destination. If the destination party becomes idle within a configurable period of time, the camped call alerts the destination party. If the camped call remains unanswered beyond the configurable period, the call recalls the transferring party.

Busy Camp On applies only to Blind Call Transfers.

3.31.1.7   Diversion Inhibitor

The Diversion Inhibitor feature allows the user to prevent calls transferred by them from being redirected by the transferred-to party. When used with Call Transfer Recall, it allows users to ensure that the call is handled by a live person (the receptionist), if the intended destination is unavailable to answer the call. It can be enabled by the user independently for blind transferred calls and for calls transferred with consultation.

3.31.2 Configuration

To initiate a transfer with Three-Way Consultation, the user must have the Three-Way Calling service assigned.

Users can configure the following for their Call Transfer service:

- Enable or disable Call Transfer Recall functionality.

- Set the number of rings before Call Transfer Recall is automatically triggered.

- Enable or disable Busy Camp On functionality and specify the time after which the transferring user should be recalled.

- Enable or disable the use of Diversion Inhibitor independently for blind transferred calls and calls transferred with consultation.

3.32 Call Waiting

The Call Waiting service enables a user to answer a call while already engaged in another call.

3.32.1 Description

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

If the user hangs up while another party in the session is held or waiting, the user is re-rung. Upon answering, the user is reconnected to the held party.

The service may optionally provide the caller with a distinctive ringback tone when the called party is already busy on a call.

The service ends when any party hangs up.

3.32.2 Configuration

The system administrator configures whether a distinctive ringback tone should be applied to inform callers when a called party is already busy on another call.

The user can activate and deactivate the service and specify whether they want the calling line ID (CLID) of the waiting caller to be delivered to their device while they are busy on another call.

The user can also activate and deactivate the service using feature access codes:

- To activate the service, the user dials *43 (default). The system then plays a confirmation announcement and the user hangs up.

- To deactivate the service, the user dials #43 (default). The system then plays a confirmation announcement and the user hangs up.

- To obtain the current status of Call Waiting, the user dials *53* (default).

In addition, the user can cancel Call Waiting just for one call, for the next call, or for the call in progress.

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

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

3.33 Charge Number

The Charge Number service allows administrators to associate an additional directory number with a user account for charging.

3.33.1 Description

The directory number may be any number available to the user's group, including numbers already assigned to other user accounts.

The selected number is recorded in all call detail records (CDRs) generated for the users' outgoing calls. Records for calls that are redirected to the network only contain the charge number of the last redirecting party, if one exists; they do not contain the charge number of the redirected party.

The Charge Number service allows a configurable URI scheme. When a Cisco BroadWorks user with a charge number provisioned originates a network call, the system includes a *Charge* header in the connection request message sent to the network depending on the system configuration.

3.33.2 Configuration

The Charge Number service is configured for the user by a group administrator.

In addition, the administrator can specify whether the user's charge number should be used for enhanced translation and/or included in network calls.

The user can view the charge number associated with their account, but they cannot modify the settings configured by their administrator.

The system administrator configures the format of the *SIP Charge* header using the CLI.

3.34 Collaborate

Collaborate – Audio and Collaborate – Video services integrate audio and video conferencing capabilities into multiuser chat rooms on the UC-One client. In addition, the Collaborate – Sharing service allows the user to use the sharing functionality of their UC-One client.

3.34.1 Description

Collaborate – Audio and Collaborate – Video services allow a user to integrate audio and video conferences into their multiuser chat rooms on their UC-One client.

There are three types of multiuser chat rooms supported by the UC-One client:

- My Room – This is a default multiuser chat room available to all UC-One clients. This room is available to all users and is advertised via the enterprise directory. It is also advertised to the user's buddies.

- Project Rooms – These are rooms created for a specific topic. These rooms can have indefinite life spans such as My Room or they can support one-time or schedulable sessions.

- Instant Rooms – These rooms are created as needed by the client. They only exist for as long as desired by the user that created the room.

Collaborate rooms on Cisco BroadWorks provide audio and video conferencing capabilities associated with those chat rooms. The UC-One client provides a seamless interface to the user and links the conference and the chat room. Participants in a collaboration session can use conferencing, chatting, or both capabilities.

Collaborate rooms also support the participation of non-UC-One clients in the audio/video conferences and chat sessions. Non-UC-One clients can access a conference through an external directory number (DN) and the room ID, or by the room owner's outdialing the client into the conference. A chat session can be entered through a web link provided to the non-UC-One client through an external means such as email or a one-to-one chat session.

A video collaboration room can handle up to 30 participants, 15 of which can be video participants. Once 15 video participants are connected to a video collaboration room, only audio participants are allowed to connect to the room.

When the Collaborate – Audio service is authorized to a group, a new default collaborate bridge is created for the group. This bridge hosts the audio and video conferences associated with a user's multiuser chat rooms on the UC-One client. Once the Collaborate – Audio service is assigned to the user, their default room, My Room, is created on Cisco BroadWorks. My Room provides default audio conferencing associated with the user's UC-One My Room multiuser chat room. The room ID is used to access the audio/video conferences in the particular room.

Users can create project rooms and configure general settings for their instant rooms, which are dynamically created by the client when needed.

The Collaborate – Video service is assigned to the user if they require video capabilities in their collaboration rooms. The user must have both Collaborate – Audio and Collaborate – Video services assigned to have video conferencing capabilities.

Collaborate – Video supports the H.264 advanced video coding (AVC) constrained baseline profile up to level 3.1, which natively supports up to 720p HD resolution.

In addition, the Collaborate – Sharing service allows the user to use the sharing functionality of their UC-One client.

3.34.2 Configuration

The system administrator configures the Video Server selection parameters and the system-level Collaborate service parameters.

■ The Video Server selection parameters define Application Server behavior for when the Application Server detects a loss of communication to the Video Server. They include the time before the Application Server advances to next Video Server, the statically-configured Video Server indication, and the provisioning of static Video Servers.

■ The Collaborate service parameters specify the collaborate bridges and room behaviors, such as, outdial support, maximum length of a collaborate room session, and maximum number of session participants.

The group administrator can define additional bridges in addition to the default bridge and assigned users (who have the Collaborate – Audio service) to it.

Users with the Collaborate – Audio service assigned can create project rooms and configure instant room common settings. They can also modify, but not delete their My Room.

3.35 CommPilot Express

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

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

3.35.1 Description

CommPilot Express is a meta-service that consolidates Cisco BroadWorks Call Termination services into four profile-based call management templates. Each profile includes preferences for managing the relevant incoming call functions (for example, Call Forwarding [Busy, No Answer, Always, Selective], Voice Messaging, Simultaneous Ringing, Call Notify), which can be configured through a single easy-to-use *web* page.

The following profiles are defined:

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

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

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

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

3.35.2 Configuration

The user configures the CommPilot Express profiles and uploads a personal greeting through the web portal.

When the user enters a 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 select their active CommPilot Express profile in one of the ways shown in the following table.

Item	Description
CommPilot Personal Web Portal	The user can use their *CommPilot Express* page on the web portal to select their active profile.
CommPilot Voice Portal	When on the road or without web access, the user can call into their voice portal to select their active CommPilot Express profile.

The no-answer timer for no-answer handling services used by CommPilot Express is not set in CommPilot Express, but in can be set in any of no-answer handling services and is shared by all no-answer handling services. For more information, see section *6.14 User No Answer Timer*.

3.36 CommPilot Personal Web Portal

This feature allows a user to configure and customize services.

3.36.1 Description

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

The CommPilot Personal web portal also provides the users with the following:

- Read access to group services to which they are subscribed
- List of feature access codes that are associated with subscribed-to services
- Context-sensitive help for every service
- Read access to their calling plans
- Access to phone directories
- Access to call logs

The CommPilot Personal web portal is authenticated for each user with a user ID and a password, and it provides a secure connection to Cisco BroadWorks.

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

3.36.1.1 Login Wizard

When a new user attempts to log in for the first time, the system detects the initial login attempt and redirects the user to a page where the user can change their password.

When a user attempts to log in with an expired password, the system also redirects the user to the same page to change the password.

In both cases, the system detects a valid user name and password. The user's identity is authenticated so that Cisco BroadWorks can provide the proper visual branding for the *Password Change* web page. However, the user is not authorized to do anything except change their password. The user is unable to navigate to any other pages within the web portal. Note that attempting to navigate to other pages fails, when typing the URLs directly into the browser address bar.

When the user successfully changes their password and upon subsequent successful logins, the user is redirected to their home page.

3.36.2 Configuration

This feature is automatically available to Cisco BroadWorks users and does not require configuration.

3.37 Communication Barring User-Control

The Communication Barring User-Control (CB-UC) service allows a user to screen calls of specific types made from their account.

3.37.1 Description

This service is typically used to prevent other users from making certain types of calls from the user's account when the user's devices are accessible to others. For example, this service may be offered to the residential market so that parents can limit the calls that their children can make.

Users select, from a preconfigured list, a barring profile to be used to screen outgoing calls.

3.37.2 Configuration

At the system level, the administrator sets the Communication Barring User-Control personal identification number (PIN) code lockout parameters.

The user selects the active profile, activates and deactivates the service, and sets their PIN using the web portal. The profiles the user can select from are those that are part of the Network Class of Service (NCOS) assigned to the user. Only one profile can be active at any given time.

The user can also activate and deactivate the service using the feature access codes from their device.

3.37.2.1 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 *33* (default). The system then plays a confirmation announcement and the user hangs up.

- To deactivate the service, the user dials #33* (default). The system then plays a confirmation announcement and the user hangs up.

- To obtain the status of the service, the user dials *#33#; and enters a PIN provided by their service provider.

  The query allows the user to find out whether the service is currently active or inactive. If the service is active, the announcement also informs the user which profile is selected.

3.37.2.2 Lockout

When enabled by the system administrator, the lockout option detects repeated attempts to access the service with an erroneous PIN and locks service activation and deactivation via the telephone interface for a configurable amount of time. The lockout has no effect on querying the service status.

While the access is locked out, an announcement is played to inform the user about the lockout. The service status remains unchanged.

The user can still access the service configuration using the web portal and can change the PIN, which resets the lockout.

3.38 Conference Room

This feature adds the Conference Room user service license to Cisco BroadWorks.

3.38.1 Description

Conference Room is a specialized application designed for small to medium-sized meeting or huddle rooms to include physical rooms into the My Room audio, video, and content experience.

3.38.2 Configuration

This service has no configuration.

3.39   Configurable Calling Line ID

The Configurable Calling Line ID feature allows an alternate calling line ID to be presented to the called party when the user makes a call.

3.39.1 Description

This feature allows a group administrator to assign an alternate calling line ID, which is delivered to the called party.

The calling line ID consists of the last name, first name, and phone number, as they should be delivered as part of the calling line ID. The user cannot modify these attributes, but can see them in their profile.

3.39.2 Configuration

This feature is available automatically and it does not need to be assigned to the user.

The group administrator configures the user's calling line ID attributes. When not defined, the user's actual name and number are used by default.

The Call Processing policies determine how the configurable calling line ID is used in calls. For more information, see section *6.3 Call Processing Policies*.

3.40 Connected Line Identification Presentation

This service allows the calling party to be presented with the identity of the connected party, which may or may not be the dialed party.

3.40.1 Description

The Connected Line Identification Presentation service is a key Integrated Services Digital Network (ISDN) service that is usually offered to primary rate interface (PRI) and basic rate interface (BRI) terminals.

This originating user service is an overlay service to the existing Internal Calling Line ID Delivery and External Calling Line ID Delivery user services. If a user is assigned the Connected Line Identification Presentation service but is not assigned the necessary Calling Line ID Delivery service for a call or the necessary service is disabled, the Connected Line Identification Presentation service is considered disabled. The Connected Line Identification Presentation service controls whether the user receives the connected identity of the remote party on their Xtended Services Interface (Xsi) clients and SIP devices.

The connected line information associated with any calling/called party is populated according to that party's effective Calling Line ID policy.

3.40.2 Configuration

Since this service is an overlay to the Internal Calling Line ID Delivery and External Calling Line ID Delivery services, the user must have the appropriate Calling Line ID Delivery service assigned to use this service. The user enables or disables the service by enabling or disabling the Internal Calling Line ID Delivery and/or the External Calling Line ID Delivery services.

3.41 Connected Line Identification Restriction

This service allows a user to control whether their connected identity is restricted when being sent to the remote user for Connected Line Identification Presentation purposes.

3.41.1 Description

The Connected Line Identification Restriction service is a key Integrated Services Digital Network (ISDN) service that is usually offered to primary rate interface and basic rate interface terminals.

This is a terminating service and does not apply to originations from the user. The user's connected identity is restricted on incoming calls.

3.41.2 Configuration

The service can also be assigned to virtual users, such as call centers or trunk groups.

The user enables or disables the service through the web portal.

The user can query the status of this service by dialing *56* (default).

3.42 Consultation Hold

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

3.42.1 Description

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

3.42.2 Configuration

This service does not need to be assigned to the user and has no configuration parameters.

3.43 Customer Originated Trace (Call Trace)

The Customer Originated Trace service enables the recipient of an obscene, harassing, or threatening call to request that it be automatically traced either by dialing a feature access code after the call or by pressing a key "mid-call".

3.43.1 Description

The user is able to trace an incoming call by dialing *57 (default) after the call is received. When the user issues a trace request, the call that was last received by the user is traced. It could be either an answered or a missed call.

After dialing a feature access code, the user hears a confirmation announcement followed by the dial tone and the requested trace is sent in the form of an alarm to the system provider.

The user can also press a key to invoke Customer Originated Trace mid-call for an answered call. When invoked mid-call, no announcements play to the user, since it is desirable not to affect the media stream between the user and the caller.

The trace contains the following information:

- The phone number of the user who initiated the trace

  When the user does not have a phone number, the group phone number and the extension of the user are provided instead. For intragroup calls, only the extension is used.

- The date and time the call was received

- The identity (name and/or number) of the caller

However, if neither the caller's name nor the caller's number is available to Cisco BroadWorks, the call cannot be traced, and the user receives an error announcement when they issue a trace request.

3.43.2 Configuration

The service has no configuration.

3.44 Custom Ringback User

The Custom Ringback User service allows a user to play a custom ringback tone instead of the standard ringback tone for calls that satisfy selective criteria defined by the user.

3.44.1 Description

This service allows a user to configure selective criteria used to filter incoming calls and to specify custom media files to be used as ringback for calls that satisfy the criteria.

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 workweek. For more information, see section *6.18 Selective Criteria*.

Multiple criteria entries can be defined and a custom media file can be associated with each criteria entry.

When a call is received, it is compared with the criteria entries created by the user. If a match is found, then the associated custom media file is used; otherwise, the group service is checked. If active, then the group's custom media file is used; otherwise, the usual ringback procedures are applied to the call.

Note that the service is automatically turned off when there are no active criteria entries.

The following related services are also available:

■   The Custom Ringback User – Video – This service allows video ringback files to be configured in each criteria entry, in addition to audio ringback files.

■   The Custom Ringback – Call Waiting – This service allows the user to configure a different ringback to play for call waiting calls.

3.44.2 Configuration

The user defines the criteria entries used to filter incoming calls. Each criteria entry specifies the selective criteria (phone numbers, time of day, holiday schedule, and so on) to be applied to incoming calls and has its own defined audio file specified.

A separate audio can be specified for initial and waiting ringback if the user has the Custom Ringback User – Call Waiting service assigned.

A video ringback can be configured in each criteria entry if the user has the Custom Ringback User – Video service assigned.

In addition, criteria entries can be selectively activated or deactivated.

> **NOTE**: For each criteria entry, at least one Custom Ringback file (audio or video) should be defined.

Custom Ringback audio and video files loaded via the web interface are subject to a length and format validation.

3.45 Dialable Caller ID

The Dialable Caller ID service allows a user to distinguish between local and toll inbound calls based on the calling number and return calls directly from their local phone call logs without having to manipulate the digits.

3.45.1 Description

A dialable caller ID is a calling number in a format that allows the user to call the number directly without having to manipulate the digits.

Upon receiving an incoming network call for a Cisco BroadWorks user, the system performs reverse translation to determine the format of the remote party's phone number, and gathers other information required to translate the number into a dialable format.

Once the calling number is translated, it is delivered to the user as a dialable caller ID. The user can then use the call log to make a call.

3.45.2 Configuration

The following settings must be configured for the service:

■ Incoming Caller ID Call Processing policy at the system, service provider/enterprise, group, and user levels

■ Dialable Caller ID Access Administrator policy for service provider/enterprise and group administrators to control administrators' access to the Dialable Caller ID configuration

■ Dialable Caller ID criteria at the system, service provider/enterprise, and group levels

NOTE: Dialable Caller ID and Communication Barring share the alternate call indicators and call types. Configuring these settings (via the CLI) affects both services.

In IMS mode, the system administrator specifies whether enhanced network translations should be used.

In addition, the system administrator configures the Reverse Translation policy and call screening rules on the Network Server.

3.46 Directed Call Pickup

The Directed Call Pickup service allows a user to pick up (answer) a call directed to another user in the same customer group (or enterprise, if the group is part of an enterprise).

3.46.1 Description

To pick up a call, the user dials *97 (default), followed by the extension of the ringing party. If the user does not supply an extension, they are given a stutter dial tone so that they can enter the extension.

When the call is answered, the user can treat the call as any other call they receive.

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

3.46.2 Configuration

This service does not have to be assigned to the user and requires no configuration.

3.47 Directed Call Pickup with Barge-in

The Directed Call Pickup with Barge-in (DPUBI) service allows a user to pick up (answer) a call directed to another user in the same customer group (or enterprise, if the group is part of an enterprise), or barge in on the call if the call was already answered.

3.47.1 Description

A user invokes the service by dialing *33 (default) followed by the extension to be picked up. If the user does not supply an extension, they are given a stutter dial tone so that they can enter the extension.

If the user at the dialed extension has not answered the call, a pickup occurs; that is, the user who invoked the service and the calling party are connected to one another, and the ringing party is released.

If the ringing party has already answered the call, a barge-in occurs; that is, a three-way call is established between the parties with the user who invoked the service as the controller. There is no limit to the number of users, who can barge in on a call. If more than one user barges in on the same call, then a cascade of three-way calls occurs, forming a larger conference. The system does not enforce a limit on that cascading pattern.

> **NOTE**: The pickup portion of the Directed Call Pickup with Barge-in service is identical to the Directed Call Pickup service. However, Directed Call Pickup with Barge-in is a completely separate service (from Directed Call Pickup) that adds the barge-in capability and has its own feature access code.

If the picked-up party has no calls or more than one call, the user who invoked the service is given a reorder tone. A pickup or barge-in can occur only when the picked-up party has exactly one call.

In the remainder of this section, the following terms apply:

- DPUBI user – The user invoking the Directed Call Pickup with Barge-in service
- Picked-up user – The user whose extension has been selected by the DPUBI user
- Other party – The party that is connected to the picked-up user when the DPUBI attempt takes place.

3.47.1.1 Barge-in

A barge-in is triggered when the user at the dialed extension has a single call, regardless whether the call was originating or terminating, and regardless of its current state (active or held).

When a barge-in occurs and the DPUBI user has the warning tone option enabled, the picked-up user is given the barge-in warning tone. The other party is put on hold while the picked-up user is receiving the warning tone.

> **NOTE**: The picked-up user does not give a warning tone if they have put the call on hold.

If the DPUBI user has the Three-Way Call service assigned and flashes during the three-way call, the other party is dropped from the call. If the DPUBI user does not have the Three-Way Call service and flashes, the flash is ignored.

If the DPUBI user hangs up, the picked-up user and the other party are transferred together. The transfer is not subjected to the screening by the DPUBI user's Outgoing Calling Plans.

3.47.1.2   Automatic Target Selection

A configurable option allows automatic target selection for either pickup or barge-in.

With this option, the user can dial the DPUBI feature access code without providing the extension, and a target for pickup or barge-in is automatically selected if, at that moment, there is only one user from the same group involved in a call (connected or ringing). This option is primarily targeted at enhancing residential multiline deployments in which a user attempts to barge in on an existing call involving another user in the household, although it can also be used in business environments.

> **NOTE**: Automatic target selection is not available across different enterprise groups. For best results, automatic target selection should be disabled for DPUBI users who need to pick up targets outside of their group.

If the target is ambiguous because multiple users meet the condition, the DPUBI user is given a stutter dial tone to enter the extension, as in the regular Directed Call Pickup with Barge-in service.

3.47.1.3   Barge-in Exempt

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

When automatic target selection is enabled and the target selected has a single active call in addition to Barge-in Exempt, the DPUBI user is provided with the DPUBI error treatment.

However, if a user has Barge-in Exempt enabled but has a single alerting call, the call can be picked up by another user using the Directed Call Pickup with Barge-in service. Barge-in Exempt does not block pickup attempts.

3.47.2 Configuration

The administrator configures the service for the user. The user can view their settings but cannot change them. The following options can be configured for the user:

■   Whether a warning tone is given to the picked-up user when a barge-in occurs

■   Whether automatic target selection is enabled

The user can activate and deactivate their Barge-in Exempt service.

3.48 Direct Inward/Outward Dialing

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

3.48.1 Description

Any Cisco BroadWorks user can be assigned a public phone number (also known as, a direct inward dialing phone number, or DID) that can be used by external parties to call the user directly, without having to go through an attendant. Cisco BroadWorks provides the required translation, routing, and location policies to terminate inbound calls to the user associated with the dialed number.

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

3.48.2 Configuration

This service has no configuration.

For a description of the use of an outside access code that is related to Direct Outward Dialing (DOD), see section *4.24 Outside Access Code*.

3.49 Diversion Inhibitor

The Diversion Inhibitor service prevents calls that are redirected by a user, from being redirected again by the called party.

3.49.1 Description

This service allows a user to inhibit redirecting services on the terminating side of an unanswered call. It is especially useful with services such as Simultaneous Ringing and Sequential Ringing. For example, if Simultaneous Ringing is engaged on a call and one of the lines has voice mail pickup set for two rings, Diversion Inhibitor continues to ring all the lines' past the two rings and prevents the call from being transferred to voice mail.

The user activates Diversion Inhibitor on a per-call basis using a feature access code as a dial prefix to the destination number. In addition, the feature access code can be configured statically as a prefix to the destination number in certain redirection services.

The following services can be inhibited with the Diversion Inhibitor feature access code:

- Call Forwarding Always, Busy, No Answer, and Selective
- Voice Mail (Cisco BroadWorks and external)
- Simultaneous Ringing (Personal)
- Sequential Ringing
- CommPilot Express

The following redirection services cannot be inhibited:

- Remote Office
- Hunt Group
- Auto Attendant
- Call Center
- Call Pickup (all variations)

3.49.2 Configuration

The service has no configuration.

3.50 Do Not Disturb

This service allows users to set their status to "unavailable" and they are not be notified of incoming calls.

3.50.1 Description

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

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

Every time a call is blocked or deflected because of the service, the user is played a ring splash as a reminder that the service is active. Users can activate and deactivate the ring splash reminder through the web portal.

3.50.2 Configuration

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

3.50.2.1 CommPilot Web Portal

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

3.50.2.2 Feature Access Code

The user can activate and deactivate the Do Not Disturb service by dialing *78 (default) to activate or *79 (default) to deactivate.

3.50.2.3 SIP Phone

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

3.51 Executive

The Executive service allows users to have a pool of assistants who can answer and initiate calls on their behalf.

3.51.1 Description

Executive and Executive-Assistant are two interrelated services that together deliver the following functionality:

- A user with the Executive service can define a pool of assistants who manage their calls. The assistants have to be selected among the users in the same group or enterprise who have the Executive-Assistant service assigned.

- A user with the Executive-Assistant service can answer and initiate calls on behalf of their executives.

- Both the executive and their assistants can specify which calls should be forwarded to the assistants, how assistants should be alerted about incoming calls, and which of the calls forwarded to the assistants should be presented to the executive for screening.

3.51.1.1 Call Filtering

Call filtering determines whether an incoming call intended for the executive should be routed to their assistants. If the call passes filtering, it is routed to the available assistants (assistants who have not opted out of the executive's pool).

Assistants can be alerted either sequentially or simultaneously about incoming calls and can be ordered for sequential alerting.

When no assistant answers the call, a rollover action is applied to the call, which can be one of the following:

- Send the call to voice mail

- Forward the call to a specified location

- No answer processing

The executive or an assistant configures which calls to forward to the executive's assistants via the web portal and can enable or disable call filtering.

In addition, the executive can activate or deactivate call filtering using feature access codes. To enable or disable call filtering, the executive dials the Executive Call Filtering Activation/Executive Call Filtering Deactivation feature access code.

When call filtering is disabled, all calls are routed to the executive.

3.51.1.2 Call Screening

Calls that are filtered and routed to the assistants can also be presented to the executive for screening. The following locations can be alerted in addition to the executive's main location:

- BroadWorks Mobility location

- BroadWorks Anywhere locations

- Shared Call Appearance locations

The executive's main location is always alerted when call screening is enabled.

If the executive answers a call, all other call legs are released.

3.51.1.3 Executive Call Initiation and Call Push

An assistant can initiate a call on behalf of their executive. To initiate a call, the assistant dials the Executive-Assistant Initiate Call feature access code, followed by the executive's number and the destination number. If any of the numbers is not provided, the assistant is prompted to provide it.

To transfer a call to the executive for whom the call is intended, the assistant dials the Executive-Assistant Call Push feature access code. The call can be either a filtered call answered by the assistant or a call initiated by the assistant on behalf of the executive via the Executive-Assistant Initiate Call feature access code.

The request is only valid when the assistant has a single answered executive call, and that answered call is connected to the same location from which the assistant dialed the feature access code. Otherwise, the call is released. If the executive does not answer the call within a configured time, the call is recalled to the assistant's device.

3.51.1.4 Call Retrieve

An executive can use the Call Retrieve feature access code to retrieve a call that is connected to an assistant's location. An assistant can use the Call Retrieve feature access code via the Executive-Assistant Initiate Call feature access code to retrieve an executive's call to the assistant's location.

To retrieve a held call from the executive's device, the assistant dials the Executive-Assistant Initiate Call feature access code and enters the Call Retrieve feature access code as the destination number. The call is retrieved from hold and the remote party is connected to the assistant's retrieving location. The call leg to the executive's device is released.

3.51.1.5 Call Bridge

An executive can use the Call Bridge feature access code to bridge to a call that is connected to an assistant location. An assistant can use the Call Bridge feature access code via the Executive-Assistant Initiate Call feature access code to bridge to the executive's call from the assistant's location.

To bridge to the executive's answered call, the assistant dials the Executive-Assistant Initiate Call feature access code and enters the Call Bridge feature access code as the destination number. The bridge warning tone is played according to the executive's configuration, and the assistant, executive, and remote party can hear each other on the bridge.

3.51.1.6 Assistant Shared Call Appearance

In some deployments, it is useful for the executive to have a Shared Call Appearance location configured for the assistant so that the assistant receives call status notifications for the executive and can directly originate, bridge, and retrieve calls on behalf of the executive without using feature access codes.

The executive's Shared Call Appearance location for the assistant should not be alerted for all incoming calls to the executive or for the executive's Click To Dial calls.

This can be set up by adding the assistant's number as the executive's Shared Call Appearance location and disabling call termination to this location.

Filtered calls still alert the assistant via executive call filtering.

3.51.1.7 Assistant BroadWorks Anywhere

The BroadWorks Anywhere E.164 Dialing feature access code is available to users with the Executive-Assistant service.

The assistant can use the BroadWorks Anywhere E.164 Dialing feature access code (*14 by default) to enter the destination address in E.164 format. The BroadWorks Anywhere E.164 Dialing feature access code must be the first digits entered for the destination number, and is replaced with a "+" to form an E.164 destination address. If the assistant enters the BroadWorks Anywhere E.164 Dialing feature access code with no additional digits, the assistant is prompted for the remaining destination address digits.

3.51.1.8 Asssitant Opt-in/Opt-out and Call Diversion

An executive can decide to allow assistants to opt in and out of their pool of assistants. When permitted, an assistant uses the Executive-Assistant Opt-in and Executive-Assistant Opt-out feature access codes to opt in or out of the executive's pool.

To opt in or out of an executive's pool, the assistant dials the Executive-Assistant Opt-in/Executive-Assistant Opt-out feature access code followed by the executive's number.

Assistants can also divert calls received on behalf of their executives to a specified location. However, the assistants' Call Forwarding and Voice Messaging services are disabled for filtered calls.

3.51.2 Configuration

3.51.2.1 Executive

A user with the Executive service can configure the following settings:

- Assistants – The executive configures the list of assistants that are assigned to the executive and specifies whether the assistants can opt in or opt out of the pool. The list of assistants is ordered for sequential alerting purposes, and all assistants must be within the executive's group or enterprise.

- Filtering – The executive enables or disables filtering and specifies which calls should be routed to assistants.

- Screening – The executive enables or disables screening and specifies the alert type and the locations to alert for screening.

- Alerting – The executive specifies whether assistants should be alerted simultaneously or sequentially, configures the Calling Line ID (CLID) name and number for filtered calls, specifies various timers and the action to take for unanswered filtered calls.

3.51.2.2 Executive-Assistant

A user with the Executive-Assistant service can configure the following settings:

- Divert – The assistant can enable or disable call diversion of filtered calls and specify the address to which filtered calls are diverted.

- Opt-in/Opt-out – For any executive that the assistant is assigned to, who allows assistants to opt in and out of their pool of assistants, the assistant can configure whether they opt in or opt out for that executive.

- Executive Settings – For any executive, the assistant is assigned to, the assistant can access and modify their filtering, screening, and alerting settings. However, the assistant cannot access or modify the executive's pool of assistants.

3.52 Explicit Call Transfer

The Explicit Call Transfer (ECT) feature allows a user with an intelligent access device to transfer a call by entering a predefined digit sequence. The need for this feature comes from the mobile space where calls are typically transferred through an explicit digit sequence (that is, *1).

3.52.1 Description

This feature monitors dual-tone multi-frequency (DTMF) events from the user's intelligent access device for the Explicit Call Transfer digit sequence and performs Call Transfer upon detecting the sequence.

The system supports Explicit Call Transfer through the interworking between the IP Multimedia Subsystem (IMS) and the Public Switched Telephone Network (PSTN) or Public Land Mobile Network (PLMN).

3.52.2 Configuration

Explicit Call Transfer is part of the In-Call Service Activation service offering and cannot be assigned or enabled separately. Enabling In-Call Service Activation automatically enables Explicit Call Transfer for the user.

The configuration of this feature includes the following:

- A system-wide DTMF string used to performing explicit call transfer

  The default value of this string is *1.

- A service provider-specific DTMF string for performing explicit call transfer

  The default value is the value of the system-wide DTMF string.

The user can view the DTMF string on their *In-Call Service Activation* configuration page.

3.53 Flash Call Hold

The Flash Call Hold service allows a user to place a call on hold and retrieve a held call by dialing a feature access code after flashing the switch hook.

3.53.1 Description

When a station user is on a two-port call and wants to place the call on hold, the user flashes the switch hook, receives a special dial tone, and dials the Flash Call Hold feature access code. The station then receives a confirmation tone to indicate that the call is being held.

The user can then make another call while the first call is held. Subsequent flashes followed by the Flash Call Hold feature access code cause the active and held calls to toggle, that is, the active call becomes held and the held call becomes active. Note that flashing the switch hook does not conference the parties but rather returns a special dial tone.

If the user hangs up after bringing in a second party following the Flash Call Hold feature access code, the two remaining parties are not connected together. Rather, the party on hold is recalled to the user's device.

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

- The holding party goes on-hook. The held call is recalled and the holding party is alerted with power ringing. When the holding party goes off-hook, the connection with the held party is restored.

- The held station hangs up, terminating the connection.

- The holding party flashes the switch hook (receives the special dial tone) and dials the Flash Call Hold feature activation code again. The holding station is then reconnected to the held party.

As is true of other Flash services, Flash Call Hold is not supported on SIP phones.

3.53.2 Configuration

This service has no configuration.

3.54 Flexible Seating

Flexible Seating allows users to use a host phone and have the host phone provisioned with the guest's device profile settings. It provides similar functionality to the Hoteling service, but allows the device to be provisioned with the guest's profile settings and has a different licensing model. It also allows the user to continue using their other locations as usual.

3.54.1 Description

Flexible Seating functionality is delivered in two separate services: Flexible Seating Host group service and Flexible Seating Guest user service.

- Flexible Seating Guest allows a user to have a device profile associated with their service. The user can then associate their profile with a flexible seating host and use the host's device with their own device profile. When a flexible seating guest uses the host's device, the device is reconfigured with the guest's device profile, so the user can have the same calling experience as if they were using their own phone. Flexible Seating Guest is a licensed user service.

- Flexible Seating Host allows group administrators to create Flexible Seating Host service instances and assign devices to them allowing users with the Flexible Seating Guest service to associate their device profile with a host. The host's phone device is allowed to download the guest's device profile. Flexible Seating Host is not a licensed service. Hosts can be created when the Flexible Seating Guest service is authorized to the group.

When the guest makes or receives a call at the host's phone, services and policies are applied to the call as if the guest were using their own phone.

3.54.1.1 Guest-Host Association and Dissociation

The guest-host association is done the same way as for the Hoteling service. For more information, see section *3.57.1.1 Guest-Host Association and Dissociation*.

> **WARNING**: The flexible seating guest line/port is only meant for use when associated with a flexible seating host and should never be associated with a real device outside of an association with a flexible seating host. Such a configuration is unsupported and causes issues when both the guest and the host devices register or subscribe using the same line/port.
>
> This restriction applies at all times, whether the guest is associated with the host or not.

3.54.1.2 Call Originations

When a guest originates a call from the host's device, the request sent from the device carries the identity provisioned for their Flexible Seating Guest service. The system identifies the host-guest association, and processes the call with the guest's service profile.

If the call is a non-emergency call, the presentation identity sent to the network or user is set based on the guest user's Call Processing policies and device profile settings.

If the call is an emergency call, the presentation number sent to the network is set based on the host's Call Processing policies.

If the host is not associated with a guest, only emergency and voice portal calls are allowed, if enabled by the host's Routing policy.

3.54.1.3 Call Terminations

A call terminated to the guest is processed with the guest's profile. If the call is not redirected by guest's services, the call is forked to the host's device (configured with the guest's device profile) and to the guest's primary and secondary locations, if provisioned.

Call terminations to the host are prohibited.

3.54.2 Configuration

Flexible Seating Host virtual users are created and configured by the group administrator.

The Flexible Seating Guest service is configured at the user level; however, some parameters can only be configured by an administrator.

3.54.2.1 Flexible Seating Host

The administrator creates Flexible Seating Host service instances and configures the following for them:

- Basic information, such as the name, calling line identity, and time zone
- An identity/device profile and addresses
- Services
- Routing policies
- The maximum time a guest user can be associated with the host, which can be unlimited
- For groups that are part of an enterprise, the access level, which indicates whether guest users within the host's enterprise or only the host's group can be associated with the host

3.54.2.2 Flexible Seating Guest

The administrator configures a device profile for the user's Flexible Seating Guest service.

The user can configure the following for their service:

- Enable or disable the service
- Provide a PIN to unlock the host device
- Select a host with whom to associate
- Restrict the length of their association time with the host to a specified number of hours (this is a required setting when the host enforces the association time limit)

3.55 GETS Calling Line ID Delivery Blocking

The GETS Calling Line ID Delivery Blocking service prevents the user's phone number from being displayed while users make Government Emergency Telecommunications Service feature code (GETS-FC) calls.

3.55.1 Description

This feature enhances the support for GETS, allowing the user to have the option to enable or disable caller ID delivery for all GETS-FC calls (GETS-FC, GETS-FC Number Translation [GETS-FC-NT], and GETS-FC Access Number [GETS-FC-AN]). This is similar to the Calling Line ID Delivery Blocking service, but applies only to outgoing GETS-FC calls.

The Calling Line ID Delivery per Call and Calling Line ID Delivery Blocking per Call feature access codes (FACs) used to override the Calling Line ID Delivery Blocking service setting for one call also override the GETS Calling Line ID Delivery Blocking setting.

3.55.2 Configuration

The user enables or disables the service via the web portal. The user can also use feature access codes to override the service setting for one call.

3.56 Group Night Forwarding

This service provides a quick way of redirecting all calls of one or more users in a group to a specified destination.

3.56.1 Description

This service allows the group administrator to redirect all incoming calls for a user to a forwarding number, which can be an Auto Attendant, a voice mailbox, and so on. The forwarding number is specified at the group level, and is the same for all users whose calls are redirected using this service.

The redirection can be activated or deactivated at the group level either manually or automatically at specified time periods (defined by a business and/or holiday schedule). In addition, the group administrator can enable or disable the forwarding individually for individual users. User-level settings have precedence over the group-level settings.

Group-level activation applies to users who have the service assigned and configured to use group settings.

3.56.2 Configuration

A system-level parameter determines whether intergroup calls within an enterprise are affected by this service when activated for a group that is part of an enterprise. If the parameter is enabled, then intergroup calls are redirected when they reach a group that has this service enabled.

The group administrator configures the service at the group level and can enable or disable the service for individual users.

The following is configurable at the group level:

- Manual activation/deactivation – When the service is manually activated, calls are redirected regardless of the business/holiday schedules.

- Automatic activation – When the service is automatically activated, calls are redirected according to the selected business and/or holiday schedules.

- Holiday and business schedules – When the service is automatically activated, the time is specified when calls should be forwarded. The group's time zone is used to determine whether a call falls within the time/holiday schedule.

- Forwarding destination – The call is redirected to this address. This can be a SIP-URI or a directory number (DN).

The following is configurable at the user level:

- Option to use group settings – When this option is selected, the service is activated or deactivated for the user, based on the settings defined at the group level.

- Manual activation or deactivation – When the service is manually activated, calls are redirected regardless of the group-level settings.

The service is not configurable by the user.

3.57 Hoteling

Hoteling allows users to associate their service profiles with devices other than their own. The service is typically used by transient employees. For example, an enterprise can set up visitor cubicles with phones that visiting employees can use with their own service profiles.

3.57.1 Description

Hoteling functionality is delivered in two separate services: Hoteling Host and Hoteling Guest.

- The Hoteling Host service allows for the designation of a particular user account as a host, allowing users with the Hoteling Guest service to associate their service profile with this account's device.

- The Hoteling Guest service allows a user to associate their profile with a Hoteling Host account and use the Hoteling Host account's device with their service profile.

When a guest makes or receives a call at the host's phone, services and policies are applied to the call as if the guest were using their own phone.

3.57.1.1 Guest-Host Association and Dissociation

A guest associates or dissociates their profile with a host via the web portal or via the voice portal.

From the web portal, the user selects an idle host and associates (or dissociates) their service profile with that host.

Using the voice portal from the host's device and their own user ID and password, the user invokes the Flexible Seating/Hoteling menu and selects the association or dissociation option.

Only one guest can be associated with a host at any given time.

A guest can also access the voice portal using their main device or a Shared Call Appearance (SCA) location while associated with a host. The guest can then use the voice portal to disassociate from the remote host.

3.57.1.2 Service Behavior

Once the association is made, the device configured for the host account is associated with the guest user. Calls made or received by the guest from the host's device use the guest's service profile.

Calls made to the host's account trigger a not-reachable condition, and services configured on the host's service profile are applied to the calls. This is similar to the host's having no assigned primary device.

The guest's original device is unreachable (unable to receive calls) and all calls placed from the guest's device, with the exception of emergency calls, are rejected.

All non-emergency calls made by the guest from the host's device use the calling line ID of the guest. However, emergency calls made by the guest from the host's device use the calling line ID of the host so that the emergency personnel have the true location of the guest.

Once a guest is disassociated from the host, the host is free to accept other associations. The guest's and host's devices are re-associated with the guest's and host's user accounts. Calls that are active while the disassociation occurs are allowed to complete as usual.

3.57.1.3 User with Hoteling Guest and Hoteling Host Services

The same user account can have both Hoteling Guest and Hoteling Host services assigned. However, it is not possible to associate the user's Hoteling Guest service with the same user's Hoteling Host service.

3.57.2 Configuration

Hoteling Host and Hoteling Guest services are configured at the user level; however, some host parameters can only be configured by an administrator.

3.57.2.1 Hoteling Host

This service is assigned to a Cisco BroadWorks user account that operates as a host for Hoteling. Typically, such users are configured with a basic set of services and do not identify an actual person within an organization. Note however, that there are no service assignment restrictions on such users. A Hoteling Host user can have a full set of services, if desired.

The administrator configures settings for the maximum time a guest user can be associated with the host. The association limit can be enabled or disabled.

For groups that are part of an enterprise, the access level indicates whether all guest users within the host user's enterprise can be associated with the host, or whether only guest users from the host's group can be associated with the host.

The Hoteling Host service can be enabled or disabled at the user level. When it is *disabled*, no guest user can be associated with that host. If a guest user is associated when this service is *disabled*, the guest is forcibly disassociated. However, if the guest is active on a call, that call is allowed to complete.

3.57.2.2 Hoteling Guest

This service is assigned to users who need to access their services from different locations. Typically, such users identify actual persons in an organization, who travel and need to make and receive calls at different locations while keeping their service profile.

The user can configure the following for their Hoteling Guest service:

- Enable or disable the service.

  When the service is disabled, the user cannot be associated with a host.

- Optionally, limit their association time to a specified number of hours.

  After the specified time expires, the user is automatically dissociated from the host.

- Select a host with whom to associate.

3.58 In-Call Service Activation

The In-Call Service Activation allows users hosted on a time division multiplexing (TDM) switch to activate Mid-Call services using DTMF digits.

This service complements the Two-Stage Dialing service providing full service support to users in TDM-overlay architecture.

3.58.1 Description

This service allows users to activate Mid-Call services via DTMF digits by establishing the media connection through Cisco BroadWorks to detect a predefined Mid-Call service trigger, and apply the Mid-Call services logic accordingly, based on the user service profile.

The call must be made to or from a non-intelligent (that is, Cisco BroadWorks-controlled) device that has TDM overlay; otherwise, the media path is not monitored for the configured digit sequence.

With most Call Control services, calls from users are compatible with the In-Call Service Activation service and can be monitored, except for N-Way Calling and Remote Office.

3.58.2 Configuration

The configuration of this service includes the following:

■ The system administrator configures a system-wide DTMF string used for activating services mid-call. The default value for this string is ##.

■ The service provider administrator configures a service provider-specific DTMF string used for activating services mid-call. The default value is the value of the system-wide DTMF string.

■ The user can enable or disable the service and view the DTMF string to use for activating services mid-call.

Enabling this service also enables the Explicit Call Transfer service. For more information, see section *3.52 Explicit Call Transfer*.

3.59 Intercept User

The Intercept User service allows a carrier to intercept calls routed to and from 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 0").

This service can also be used to suspend service for roaming users temporarily.

The Intercept User service intercepts calls for individual users. The related Intercept Group service intercepts calls for all users in a group.

3.59.1 Description

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

The caller may hear an announcement or the call can be routed to another number or to the user's voice mail. This announcement plays back a new destination number to the caller and offers the caller to connect to this new number. This announcement can (optionally) be in video format if the caller's equipment supports video.

The system administrator can define a list of phone numbers allowed to place calls to an intercepted user, such as the service provider's customer care number.

In the partial intercept scenario, the user can receive calls; however, the user's 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 partially intercepted users. 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 BroadWorks 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.

3.59.1.1 Emergency Intercept

When a user is roaming outside the geographical area served by the carrier, emergency services may be limited.

To remedy this situation, the carrier can restrict emergency and repair calls for users with the Intercept User service, through a system-level parameter. When this parameter is enabled, the Intercept User service intercepts all emergency and repair calls made by the users, in addition to any other calls blocked by this service.

A carrier can thus suspend the service for a roaming user temporarily by activating the Intercept User service for that user. When the user returns to the serving area, the carrier can deactivate Intercept User, restoring the regular service for the user.

3.59.2 Configuration

The administrator configures the service for the user. The configuration includes the following:

- Select 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. In addition, mobile inbound calls can be allowed or intercepted independently of the selected option.

- Select outbound calls to intercept. All calls can be blocked, calls can be routed to a configurable phone number, or local calls can be allowed. In addition, mobile outbound calls can be allowed or intercepted independently or the selected option.

- Specify the treatment to apply to allowed incoming calls:

  - Parallel ringing to the intercepted user's alternate network locations can be enabled or disabled.

- Specify the 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.

- Specify the treatment to apply to intercepted outgoing calls:

  - The intercepted user can be played an out-of-service announcement.

  - The intercepted user can be routed to a configurable number after hearing the intercept announcement.

  - A partially intercepted user can be configured to hear a different announcement when they place a call.

- Optionally, replace the default announcements with custom announcements.

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

3.60 Last Number Redial

The Last Number Redial service allows a user to make a call to the last dialed number without having to reenter the number.

3.60.1 Description

To make a call to the last dialed number, the user dials *66 (default). The feature access code is replaced with the digits used for the last call that was made by the user and the call is originated as usual. The digits are obtained from the last entry in the call log for placed calls.

Furthermore, the service can be used on the original or add-on call leg (that is, after flashing the switch hook).

3.60.2 Configuration

This service has no configuration.

3.61 Legacy Automatic Callback

The Legacy Automatic Callback service allows a user hosted on Cisco BroadWorks to camp on a busy called party, including a called party hosted on a Nortel CS 2000 softswitch, and to be called back automatically when the called party hangs up.

3.61.1 Description

To invoke Legacy Automatic Callback toward the last dialed number, the user dials *96 (default). When the number becomes available, the user is automatically called back.

Cisco BroadWorks implements both the originating and terminating switch behavior. Thus, a subscriber hosted on Cisco BroadWorks can use the Automatic Callback service to reach a subscriber hosted on a Nortel CS 2000 softswitch, and vice versa. A caller on Cisco BroadWorks can also use Legacy Automatic Callback to reach another subscriber hosted on Cisco BroadWorks.

A user can have more than one callback request active at a time, up to a configurable limit. Similarly, there can be more than one request active on a busy terminating party, up to a configured limit.

A user can dial #96 (default) to cancel outstanding requests at any time before the requests complete.

3.61.2 Configuration

The system administrator can configure the maximum number of originating and terminating Legacy Automatic Callback requests and various timers used by the service.

The user can activate and deactivate the service through the web portal.

3.62 Location-Based Calling Restrictions

This service provides support for defining calling restrictions based on the location of a mobile user. Location information is obtained by inspecting *Customized Applications for Mobile Network Enhanced Logic (CAMEL)* headers in the SIP INVITE message.

3.62.1 Description

The Location-Based Calling Restrictions (LCR) service is a prerequisite for enabling zone and roaming capabilities and restrictions.

The service introduces the notion of zones (groupings of physical location of cells), which are organized into zones. A single zone in the user's office zone is designated as the user's primary zone.

When users are associated with zones, their office zone and their primary zone information are included in accounting records and can be used for billing.

Office zone information can also be used as criteria for network translations and communication barring.

Users can also configure their profile to redirect calls to voice mail when they are outside their primary zone.

3.62.2 Configuration

Zones are defined at the system level and are assigned to enterprises and service providers, and then to groups.

The group administrator assigns and configures the user's office zone and primary zone.

Using the web portal, the user can specify whether to redirect their calls to voice mail when the user is outside their primary zone.

3.63 Malicious Call Trace

The Malicious Call Trace (MCT) service allows the system provider to trace calls to and from a Cisco BroadWorks user. Specified calls trigger the generation of a report (or trace) that contains information about the calling party (number, name), the time and date the call was received, and other relevant information (for example, redirection information).

3.63.1 Description

This service enables the tracing of calls for selected users. All incoming calls, all answered incoming calls, or all incoming and outgoing calls can be traced for a given user.

An announcement may inform the calling party who is attempting termination on a user with Malicious Call Trace that the call is being reported. The caller can either disconnect and not be reported or stay connected and be reported.

The time when the report is generated depends on the trace type, and is as follows:

■　**Answered** – The report is generated when the call is answered by the user being traced or by the location to which the call was forwarded by the user being traced.

■　**Alerting** – The report is generated when the call attempts to terminate on the traced user, before any terminating services of the user are allowed to proceed.

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

The report contains a subset of the call detail record information. It contains the information that is available at the time that the report is generated.

3.63.2 Configuration

> **NOTE**: Malicious Call Trace cannot be added to a service pack.

Only a system administrator can manage the service (authorize, assign, and configure) and view the service information.

The system administrator assigns the service to the user and configures the service for the user as follows:

■　Activates or deactivates the service

■　May decide to trace calls for a specific time period only

■　Specifies whether all calls, all incoming calls, or all incoming and outgoing calls should be traced

The system administrator can view a list of all Malicious Call Trace assignments on the system via the web portal.

The system administrator also sets a system-wide parameter to enable or disable the announcement being played to callers. The parameter can be overridden by service providers. This is the only parameter service providers can see. In particular, they have no access to the list of users whose calls are being traced.

3.64 Music/Video On Hold User

The Music On Hold User service enables a user to play custom music to callers when the call is held or parked. It is used in conjunction with the group Music On Hold service and it requires the group Music On Hold service to be assigned to the group.

3.64.1 Description

The Music On Hold User service allows a user to specify the custom music source to be played to callers of held or parked calls. In addition, alternate call settings can be specified for internal calls. Users can toggle between the group and custom audio source. In addition, when the Video On Hold user service is also assigned, users can specify a custom video source.

If Music/Video On Hold is turned on, then callers hear music for calls that are held and parked; otherwise, they hear silence.

A user can turn off the music (video), for the duration of the call by dialing a feature access code prior to dialing the call or when the call is in progress. Music (video) is turned off until the user's call is disconnected. Once the call is disconnected, the Music On Hold configuration is returned to the persistent state.

- Before the call, the user turns off music (video) on hold by dialing *60 (default). The user is provided with a confirmation tone followed by a dial tone, after which the user dials the outgoing telephone number.

- When a call is in progress, the user can turn off music (video) by flashing the switch hook and then dialing *60 (default). The system provides a confirmation tone followed by the dial tone. The user can now flash back to the active call.

This service is not available to virtual users and it is not applicable to users within a call center.

3.64.2 Configuration

The Music On Hold User service requires the Music On Hold service to be assigned to the group, and the Video On Hold user service requires the Music On Hold User service to be assigned to the user.

The user activates and deactivates Music/Video On Hold via the web portal (persistently) or via a feature access code (per call). The user configures the Music/Video On Hold source via the web portal.

When neither the Music On Hold nor the Video On Hold service is assigned to the user, but the Music On Hold or Video On Hold service is assigned to the group, the user can turn the group-level Music On Hold on or off.

3.65 Number Portability Announcement

The Number Portability Announcement service provides the originating user with a portability warning announcement before allowing the call to continue.

3.65.1 Description

Number Portability has become an important part of voice service delivery. Number Portability is a regulatory service that is provided among carriers serving a specific country or region. Customers are able to change their serving carrier while retaining their number. Customers do not know whether a number they are calling has been ported. Some carriers provide special rates for calling "on-net" or within their own network. For this reason, regulatory requirements in many countries require an announcement to be played prior to terminating the call or barring the call from completing when calling a number that is not served on the originator's network.

When enabled, the Number Portability Announcement service plays an announcement to the user when they are calling a ported number, which may result in additional charges. The user then has the option to continue the call or hang up.

3.65.2 Configuration

The system administrator may configure number portability statuses and related announcements.

3.66 N-Way Calling

The N-Way Calling service provides a user with the ability to add other users to a call in progress, similar to the Three-Way Calling service, up to a maximum number configured at the system or service provider/enterprise level.

3.66.1 Description

Users can create a conference using their phones by adding other users to a call in progress. Users must have SIP devices that support the REFER method to use this service.

When the user who created a conference releases the call, all participants, and the conference are released.

For information on using the N-Way Calling service, see section *3.86 Three-Way Calling*.

Because an n-way call cannot be established using flash, the In-Call Service Activation service cannot be used to initiate an n-way call. The Three-Way Calling service must be used instead.

3.66.1.1 Video Conferencing

The N-Way Calling service supports up to 15 video participants. Of the connected video participants, at most six are displayed in a tiled layout. The same tiled layout is presented to all video participants connected to the conference. The following image shows an example of a screen layout when there are six participants in the conference. The current speaker is always displayed in region number 1.
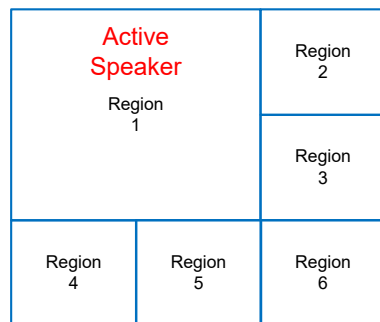
Figure 1 Video Conference – Tiled Layout with Six Participants

3.66.2 Configuration

The system administrator configures the maximum number of participants allowed in an n-way call and the URI used to identify a request for a conference resource (this setting is used for both Three-Way Calling and N-Way Calling services). The URI can be configured both at the system level and at the service provider/enterprise level. If configured at the service provider/enterprise level, it takes precedence over the system-level setting.

3.67 Office Communicator Tab

The Office Communication Tab service provides users with access to certain Cisco BroadWorks features from their Microsoft Office Communicator.

3.67.1 Description

This service adds a Cisco BroadWorks feature control tab to Microsoft Office Communicator.

3.67.2 Configuration

The service has no configuration.

3.68 Personal Assistant

The Personal Assistant service allows a user to provide an announcement to callers when the user is busy or away, specifying why they are not available to take calls and optionally when they will be back.

3.68.1 Description

The user can indicate that they are not available to take their calls by setting their presence status to one of the predefined values. The intention is to help the caller make a better decision on whether to call back, be transferred to an attendant, or leave a voice message, thus reducing the number of voice messages the user receives and allowing callers to have a better calling experience.

The possible states are as follows:

- None (equivalent to disabling the service)
- Business Trip
- Gone for the Day
- Lunch
- Meeting
- Out of the Office
- Temporarily Out
- Training
- Unavailable
- Vacation

When the user sets their presence state to any value other than "None", callers hear an announcement informing them of the reason the user is not answering the calls and optionally when they will be available. The caller can leave a message or optionally be transferred to an assistant, if the user has configured this option.

The user can specify a list of callers allowed to bypass the service and be let through. For those callers, this is equivalent to disabling the service. Other services may still apply.

In addition, the user can indicate to which of the user's devices the service should apply, by selecting them from the list, which may include the following:

- Primary number
- Alternate numbers
- BroadWorks Mobility numbers

The user can also specify whether a ring splash should be played when a call is forwarded to the assistant or to voice mail.

The user has the option to be alerted first before the caller is played the presence announcement. The announcement for the enabled presence status is played to the calling party, if the call is not answered within the user's configured maximum number of rings.

3.68.2 Configuration

The system administrator configures the keys to use to transfer a caller to voice mail and to transfer the caller to the attendant.

The user can use the web portal or the voice portal to enable or disable the service. Other settings, such as the expiry date (specifying when the user will be available), transfer to attendant, exclusion list (list of callers allowed to bypass the service), numbers to which the service applies, ring splash, the option to alert the user first, and the number of rings can only be configured through the web portal.

Note, however, that the number of rings setting applies to all services with no-answer handling. For more information and for the list of services that share this setting, see section *6.14 User No Answer Timer*.

3.69 Physical Location

The Physical Location service controls whether originating calls are allowed from physical locations other than those configured for the user's device. This provides proper support of emergency calling in countries and regions where the location of a user cannot be derived from a user's phone number.

3.69.1 Description

This service allows the restriction of the user's originating calls to the physical location configured for the user's device.

When a user originates a call, the system compares the physical location of the call (specified in the *P-Access-Network-Info* header) with the physical location configured for the user's assigned device.

- If the originating device has a configured physical location, it must match the physical location of the call exactly (except for white spaces and case) for the origination to be allowed to proceed. If the values do not match, then the origination is blocked and treatment is played. Note that all originating calls (including emergency and repair calls) are blocked.

- If the originating device has no configured physical location, then the origination is allowed to proceed whether or not the information regarding the physical location of the call is available.

3.69.2 Configuration

The system administrator can configure the service to allow emergency calls to go through when Physical Location would otherwise block them.

The administrator can enable or disable the service for the user. Users can view whether the service is enabled or disabled, but cannot modify the setting.

To restrict the calls to the physical location of the user's device, the administrator configures a physical location by editing the device profile.

3.70 Polycom Phone Services

The Polycom Phone Services user service integrates Cisco BroadWorks services with features and capabilities on the Polycom family of phones. In particular, it introduces the Polycom Phone Directory, which allows the user to add, delete, and synchronize their contacts with their Polycom phone.

3.70.1 Description

This service allows the user to select entries in their Cisco BroadWorks directory and add them as contacts to their phone directory. The phone directory can be downloaded from Cisco BroadWorks to the phone. This functionality is called Polycom Phone Directory.

3.70.2 Configuration

The group administrator can specify contacts that all users in the group should see on their Polycom phones by including them in the group's Polycom Phone Directory. It is shared by all users who have been assigned the Polycom Phone Services service, and have enabled the Polycom Phone Directory.

The group administrator configures the Polycom Phone Directory for the group by including the group's Common Phone List and/or by choosing a group's custom contact directory to include in the phone directory.

A user can enable and disable the Polycom Phone Directory on their phone. This directory includes all the contacts in the Polycom Phone Directory set up by the administrator. The user can also personalize the directory by including their personal phone list and/or by choosing a custom contact directory to include in the Polycom Phone Directory.

3.71 Pre-alerting Announcement

The Pre-alerting Announcement service allows a user to play an audio or video announcement to callers before the call is connected. Users can specify the message to be played and the conditions under which the message is played (the time, the phone numbers, and so on).

3.71.1 Description

This service allows users to configure an announcement to be played to the calling party before the user's phone is alerted.

The user also specifies the selective criteria under which the announcement is played. If the incoming call does not satisfy the conditions specified by the user, usual call handling applies.

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. If required, multiple entries can be defined and used to screen calls. For more information, see section *6.18 Selective Criteria*.

The service cannot be turned on when there are no active criteria entries.

The caller can be allowed (or not allowed) to interrupt the announcement and proceed to alerting.

A pre-alerting video announcement can also be configured and played to the caller if the caller supports the same video codec as the codec used in the video announcement. Otherwise, the pre-alerting audio announcement is played (if configured).

The media path is provided to the calling party using early media. Once the pre-alerting announcement finishes playing or the caller interrupts the announcement, the media resources are released, the terminating device is alerted, and the caller is alerted. When the terminating device answers, the two parties are connected as usual. If the Media Server fails to play the pre-alerting announcement for any reason, the terminating device is alerted as usual.

3.71.2 Configuration

The default settings for the Pre-alerting Announcement service are configured at the group level (for service provider groups) and at the enterprise level (for the enterprise groups). Enterprise and group administrators can define a default audio and video announcement for the service and specify whether the caller is allowed to interrupt the announcement and how. (A DTMF sequence to interrupt the announcement is configurable, including the option to accept any key.)

The user who has been assigned the service can:

- Activate or deactivate the service.

- Specify the audio and/or video announcement to be played.

- Define and selectively activate/deactivate criteria entries to apply to incoming calls.

3.72 Prepaid

This service allows a user to review credits from an Online Charging System (OCS) before allowing a call to proceed, reporting used credits to the OCS, and terminating a call should credits eventually run out during the call.

3.72.1 Description

The Prepaid service allows a user to make or receive calls with credits paid in advance and to view used credits.

A low credits warning tone is provided to the user a configurable number of seconds before credits run out. The warning tone informs the user that they should end the conversation quickly before being interrupted by the system.

The user can already be out of credits before a call (incoming or outgoing) is started. In this situation, treatment is provided to the caller before the called party has time to ring.

If the call is still alerting when credits run out, the called party stops ringing and the caller is provided treatment. However, if the warning-tone threshold occurs while a call is alerting or less than five seconds after the call has been answered, the warning tone is delayed until five seconds after being answered. The five-second value is hardcoded and cannot be changed.

3.72.2 Configuration

NOTE: The Prepaid service cannot be added to a service pack.

No per-user configuration is necessary apart from assigning and enabling the service to the user.

3.73 Priority Alert

The Priority Alert service enables a user to have a certain incoming calls ring on their phone with a different tone, for example, when they receive a call from their manager or spouse or when the call is from inside the group.

3.73.1 Description

This service allows a user to have certain incoming calls alert them with a different tone depending on selective criteria specified by the user. The service applies to power ringing and alerting tones. In both cases, incoming calls that meet the criteria result in a distinct ringing cadence and alerting tone pattern, respectively. The distinctive alerting pattern is the same for ringing and tones. Apart from the distinctive alerting pattern, this service does not change the way incoming calls are processed.

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. If required, multiple entries can be defined and used to screen calls. For more information, see section *6.18 Selective Criteria*.

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

The service can also be assigned to hunt groups and call centers, in which case, the analysis of the incoming call against the set of criteria is done at the hunt group or call center level, and it affects the power-ringing pattern of all agents in the group. The Priority Alert feature does not need to be assigned to the agents themselves.

3.73.2 Configuration

The user can define and selectively activate/deactivate criteria entries to apply to incoming calls.

3.74 Privacy

The Privacy service allows a user to exclude themselves from the group and enterprise directory listings as well as from the Auto Attendant extension and/or name dialing.

The user can also prevent their phone status from being available to other users.

3.74.1 Description

This service allows users to prevent the Call Center, Receptionist, and the web portal from including their user name in any group or enterprise directory lists.

Users can control whether they can be reached by name dialing and/or extension dialing and whether their phone status is being exposed to other users.

Users can also select members in an enterprise or group who are allowed to monitor their phone status. These selected members can view the user's phone status even if the user's phone status privacy is enabled.

3.74.2 Configuration

Using the web portal, users can configure the following settings for their Privacy service:

- Enable directory privacy
- Enable Auto Attendant extension and/or name dialing
- Enable phone status privacy
- Select users allowed to see their phone status when status privacy is enabled

3.75 Push To Talk

The Push To Talk (PTT) service allows a user to call another station, where the system requests that the destination station automatically answer. This provides for intercom-like functionality. The user at the destination station can control which users are allowed to call them in this way.

3.75.1 Description

This service allows users to call each other and have the call answered automatically.

A user can specify an accept list or a reject list to be used to screen incoming Push To Talk sessions.

- The accept list indicates which users are allowed to call the station.
- The reject list indicates which users are not allowed to call the station.

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

3.75.1.1 Push To Talk Origination

A user originates a Push To Talk call by dialing the Push To Talk feature access code and entering a phone number. If the user does not supply a phone number, they are given stutter dial tone so that they can enter the number to call.

> **NOTE**: The Push To Talk origination is processed by the user's origination services such as the Outgoing Calling Plan.

When the originator call has been answered, a Push To Talk confirmation tone is played to both the originator and terminator. Once the confirmation tone has finished playing, the media path between the originator and the terminator is established according to the outgoing connection type, which can be a one-way connection or a two-way connection.

If the user has a one-way outgoing connection, then no media can be transmitted from the terminator to the originator, after the call is answered. Only the originator is allowed to transmit media. If the terminator answers the call to play treatment, the originator does not hear the treatment since the call has been answered and the connection is now one way.

> **NOTE**: Before the call is answered, there is always a two-way connection so that remote media (such as remote ringback and early treatment) can be heard by the originator.

If the user has a two-way connection, then the originator and terminator can talk to each other as usual.

3.75.1.2 Push To Talk Termination

When a user receives a Push To Talk call, the call is screened using their access list, which can be an accept list or a reject list.

- If the user has configured an accept list, then the Push To Talk call is accepted only if the originator is on the list.

- If the user has configured a reject list, then the Push To Talk call is accepted only if the originator is not on the list.

When the user's access list allows the call, then the call is allowed to continue. Otherwise, the call is rejected and the originator is played an announcement.

Note that the access list can only contain other users in the group and/or enterprise. If the terminator has configured an accept list, then all Push To Talk calls from outside the group and enterprise are rejected since the originator cannot be on the list. Similarly, if the terminator has configured a reject list, then all Push To Talk calls from outside the group and enterprise are accepted since the originator cannot be on the list.

The call is automatically answered if Auto-Answer is enabled at the destination and if the destination device is compliant with Advanced Call Control (ACC). Otherwise, the Push To Talk call must be manually answered.

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

3.75.2 Configuration

Users can configure the following settings for their Push To Talk service.

- The outgoing connection type

- The Auto-Answer (on or off)

- The access list type and the access list, that is, the list of users within the same group/enterprise from whom to accept/reject Push To Talk calls

3.76 Remote Office

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

3.76.1 Description

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

To use the service, the user simply enters the phone number of their current location and activates the service. From that point on, their usual Cisco BroadWorks location is temporarily overridden by the newly configured location.

When the service is active, incoming calls to the user are redirected to their remote phone number and are subjected to the user's terminating services.

Similarly, the user can originate calls from their remote phone number using Click To Dial functionality. This ensures that calls are processed by the system as normal originating calls, and are subjected to the users' originating services.

This service allows users to manage active calls as usual through the UC-One Communicator, thus providing the users with their Cisco BroadWorks profile and services from any addressable phone on the network or the Public Switched Telephone Network (PSTN).

3.76.2 Configuration

The following table describes the configuration items for Remote Office.

Item	Description
CommPilot Personal	A user can configure their temporary location and activate the service through the CommPilot Personal web portal. The location is entered as a phone number.
UC-One Communicator	The user selects *Remote Office* and then *Configure* from the *Calls* tab and then can configure their temporary location and activate the service in the dialog box that appears.

When the user enters a 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.

3.77 Security Classification

The Security Classification service allows users to be aware of the maximum level of classified information that can be safely exchanged in a conversation.

3.77.1 Description

The Security Classification service allows the system to classify calls that a user is involved in with a security classification level. The purpose of the service is for users to be conscious of the maximum level of classified information that can be safely exchanged in a conversation. This service is only intended to notify the user of the security classification of calls.

The Security Classification service allows assigning a security classification level to a user. Security classification levels are defined at the system level and prioritized.

The security classification level for a two-way or conference call is determined as the lowest of the security classification levels for all users in the call. When the security classification level is determined, it is communicated to the user's device for display to the user.

The user can modify their assigned security classification level (via a supported device or client application) to a value lower than their assigned level while in an active call. If this modification affects the current security classification level for the call, then this value is recalculated and displayed to the users on the phone/client device.

When the user transfers a call, or when new parties join or leave a conference, then the security classification level is re-evaluated for all the parties connected. For Meet-Me conferences, an announcement is played to the bridge for the initial conference security classification level, and then whenever the conference security classification level changes.

3.77.2 Configuration

The system administrator configures system-wide security classification levels and sets the timer for controlling the wait time for playing security classification announcement in Meet-Me conferences.

In the centralized (Amplify) architecture, the system administrator can enable and disable security classification customization for the enterprise and service provider group. When security classification customization is enabled, the enterprise and service provider group administrators can customize the system security classifications for their organization.

The group administrator assigns security classification levels to users and default security classification levels to trunk groups. The group administrator also configures the users' Device policy to specify whether the Security Classification service is synchronized on shared devices.

3.78 Selective Call Acceptance

This service enables a user to accept calls selectively, based on configurable conditions. Calls that do not meet the specified conditions are rejected and provided a treatment.

3.78.1 Description

Selective Call Acceptance allows a user to accept only those calls that meet selective criteria defined by the user. The accepted calls are allowed to terminate as usual whereas the rejected calls are provided a treatment.

The criteria are based on the incoming caller identity, ranges of digits, time of day, and day of the week, and are combined into criteria entries, for example, an incoming call from this number, within business hours, and the during workweek. If required, multiple entries can be defined and used to screen calls. For more information, see section *6.18 Selective Criteria*.

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

The Cisco BroadWorks no-charge treatment capability can be used with this service.

3.78.2 Configuration

The user can define and selectively activate/deactivate criteria entries to apply to incoming calls.

3.79 Selective Call Rejection

This service enables a user to define criteria that causes certain incoming calls to be rejected and provided a treatment. Additional criteria are automatically created by the system when calls are reported as malicious by the Customer Originated Trace service. All other calls terminate as usual.

3.79.1 Description

Selective Call Rejection allows a user to block calls that meet selective criteria defined by the user. The rejected calls are provided a treatment whereas the accepted calls are allowed to terminate as usual.

The criteria are based on the incoming caller identity, ranges of digits, time of day, and day of the week, and are combined into criteria entries, for example, an incoming call from this number, within business hours, and during the workweek. If required, multiple entries can be defined and used to screen calls. For more information, see section *6.18 Selective Criteria*.

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

The Cisco BroadWorks Application Server's no-charge treatment capability can be used with this service.

3.79.2 Configuration

The user can define and selectively activate/deactivate criteria entries to apply to incoming calls.

The criteria entries created by the system for malicious calls can be deleted by the user but cannot be modified.

3.80 Sequential Ringing

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

3.80.1 Description

When the Sequential Ringing service is triggered on an incoming call, it takes control of the call and provides the calling party with an announcement stating that the system will attempt to locate the user.

The service then attempts to call the user by calling the phone numbers or URLs on the Sequential Ringing list (starting with the user's base location, if this option is enabled) one after the other until the call is answered or the last number is attempted.

If the user has set the number of rings before no-answer processing to "0", the base location is not rung, and the service proceeds to the next location.

While the service goes through the list of locations, the caller is provided with the initial greeting followed by local ringback. Since the time necessary to find the user can be considerable, a comfort announcement is periodically played to the user.

Note that nothing prevents the user from entering the same location twice or from entering their own number, for example, should the user want their base location to be alerted last.

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

- A call is originated to the phone number or URL and a timer is started. The timer is configured separately (in number of rings) for each location. The user can also set the number of rings for the base location by dialing the No Answer Timer feature access code (*610 by default). Note however, that the timer for the base location applies to all services with no-answer handling, independently of how it its set. For more information and for the list of services that share this setting, see section *6.14 User No Answer Timer.*

- If the called number is busy or results in a local announcement, the call is released and the service moves on to the next number. The service can be configured to not search any further location if the base location is busy, unreachable, or unregistered, or if there is any other reason preventing the system from alerting the location. In this context, "busy" is used as a generic term, which means that the system is unable to alert the location. If this occurs, busy processing takes place immediately.

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

- If the service is configured to allow the caller to skip the search process and the caller presses the # key, the search process is interrupted and the caller is presented with no-answer processing immediately.

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

Once a call is successfully connected, or when the last location on the list remains unanswered, the ringback or announcement is interrupted and the caller is connected to the user, or provided with no-answer processing, as applicable. If all locations are busy, busy processing is applied to the call.

The user also defines a set of criteria that determine whether the locations on the Sequential Ringing list should be attempted for the incoming call.

The criteria are based on the incoming caller identity, ranges of digits, time of day, and day of the week, and are combined into criteria entries, for example, an incoming call from this number, within business hours, and during the workweek. If required, multiple entries can be defined and used to screen calls. For more information, see section *6.18 Selective Criteria*.

If the criteria do not match, the call continues as if this service was turned off. In addition, if there are no active criteria entries, the service is automatically turned off.

The user can enable answer confirmation for each location, which allows Sequential Ringing to prompt the called party to enter a digit to confirm the acceptance of the call. Alerting all destinations until a confirmation digit is received, ensures that all of the called party's numbers are attempted before the call is rolled over to a voice mailbox.

3.80.2 Configuration

The user can configure the following:

- Whether to use the base location

- Whether to continue searching if the base location cannot be alerted

- Whether the caller can press # to interrupt the search process

- The list of up to five locations and their timers (number of rings)

    When the user enters a 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.

- Selective criteria entries and whether each one is active

- The number of rings defining the no-answer condition, which is set by dialing *610 (default) or by configuring the *Number of rings for Base Location* setting.

    Note however, that this setting applies to all services with no-answer handling. For more information and for the list of services that share this setting, see section *6.14 User No Answer Timer*.

3.81 Shared Call Appearance (Multiple Appearance Directory Numbers)

The Shared Call Appearance (SCA) service allows an incoming call to appear at multiple locations simultaneously. All devices where call appearance is shared can be used to answer an incoming call or originate a call on behalf of the primary location, such as for instance, in an administrative assistant/executive scenario.

The service also has private hold capability. When a location puts a call on private hold, only the location that held the call can retrieve it. Retrieve attempts from all other locations are rejected.

3.81.1 Basic Functionality

This service allows up to 35 locations (devices) to share the same line appearance, so that these locations all behave as extensions of a single line or user.

One of the locations is the user's primary location while the other locations are called secondary locations. Any type of device can be used for primary and secondary locations.

When a user with this service is idle, an incoming call alerts all the assigned locations. The first location to answer is considered the active location and the other locations are released and locked for further use by that user.

When the Shared Call Appearance service processes an invitation containing the *Call-Info* header with *answer-after* parameter, then it proxies it to the main location and not to any of the alternate locations.

If the user is already busy on a call on one of the locations associated with the line, the incoming call is presented to the active location only.

The user can originate a call from any location. This locks out the other locations for the duration of the call. Any further incoming calls to this user are presented to the active location, which is similar to a terminating scenario.

For Cisco BroadWorks-controlled devices, calls can be transferred between locations by hanging up on a held call. This causes all locations to be rung again, and any location can then answer the call, thus transferring the call between the two locations.

3.81.2 Location Control

Users can enable or disable individual secondary locations via the web portal or by using a feature access code. To enable a location, the user dials *12 (default) from the location to enable; to disable a location, the user dials *13 (default) from that location.

Note that feature access codes for location control are shared between BroadWorks Anywhere and Shared Call Appearance.

Disabling a location has the same effect as preventing call termination and has no effect on the ability of the user to originate calls or retrieve calls from that location. Message waiting and call information notifications continue to be sent to a location that is deactivated.

In addition, administrators can prevent call originations from a secondary location or call termination attempts to a secondary location.

3.81.3 BroadWorks Mobility

This BroadWorks Mobility service enables the user to use Cisco BroadWorks enhanced services from a mobile device.

The BroadWorks Mobility feature also runs as an overlay to the Shared Call Appearance service. Shared Call Appearance alerts the mobile whenever there is a mobile termination and the BroadWorks Mobility service is set to ring the mobile phone. The user's alternate locations are alerted in parallel to the user's mobile devices. However, if the user is already on an active call from or to the primary device, then incoming calls are only presented to the primary device, independent of the alerting policy set for the mobile.

3.81.4 Multiple Call Arrangement

The Multiple Call Arrangement (MCA) service, which is an overlay to Shared Call Appearance, provides the ability for multiple calls to be handled concurrently on different Shared Call Appearance locations of the same user. No single location is considered the active location, so all locations are allowed to originate calls and receive new incoming calls when they have an available call appearance, regardless of the activity at the other locations.

Shared Call Appearance allows a user's primary and secondary locations to be a combination of any type of devices. However, Multiple Call Arrangement is only supported with intelligent devices, such as SIP phones. Intelligent devices are devices that can provide their own call control capability including call transfer, three-way calling, and so on. If a user is provisioned with a non-intelligent device for any of their locations then Multiple Call Arrangement is disabled for this user. In addition, if any device assigned to a user, such as Windows Messenger, is controlled by Cisco BroadWorks, the system processes all incoming and outgoing calls as if the Multiple Call Arrangement service were not assigned.

3.81.5 Simultaneous Calls on Primary and One Alternate Location

In systems that work in XS mode, when the user does not have Multiple Call Arrangement or has it disabled, the administrator can enable a capability (Simultaneous Primary and Alternate Location Calls) that allows the user to have simultaneous calls on the primary and **one** alternate location. This also allows user's alternate locations to be alerted when the user is already on a call at the primary location. Depending on how this feature is configured, the user may or may not be allowed to use an alternate location to retrieve or barge in on a current call on the primary location. Only the administrator has access to those settings.

3.81.6 Call Retrieve

The Call Retrieve feature access code allows a user to dial *11 (default) from a shared location to retrieve an existing active call from another shared location.

The user can invoke the Two-Stage Dialing service to dial the feature access code.

The criteria for selecting the call to retrieve are as follows:

- If the location is involved in only one other call, then the call is retrieved if it is active, held, or remotely held.

- If the location is involved in multiple calls, then the call to retrieve must be active and it must be the only active call.

If a call satisfies one of the criteria, then the location is released and the user who dialed the feature access code is connected to the calling party. Otherwise, the user is given a reorder treatment.

> **NOTE**: Call Waiting is required with Multiple Call Arrangement to retrieve a call.

3.81.7 Hold/Retrieve

This capability allows for calls put on hold at one location to be retrieved from any other location sharing the same call appearance.

After a call is put on hold, all Shared Call Appearance devices show that call as being held by flashing the corresponding lamp slowly. At this point, any of the devices can be used to retrieve the held call. After the call is retrieved by a location, all lamps go on (solid) showing the call as active, and the location becomes the active location.

A user can retrieve a held Shared Call Appearance call by selecting the call and pressing a feature key (for example, Retrieve) on a SIP phone.

3.81.8 Call Bridge (Barge-in)

This capability allows Shared Call Appearance locations to barge in on an active call involving another location.

When one of the Shared Call Appearance locations has an active call, the devices at all locations show that that call is active. If bridging is enabled, any device can be used to barge in on that call.

A configurable option allows the user to enable or disable bridging between locations, thus enhancing the Cisco BroadWorks executive/administrator solution to fill a functional gap that exists with some Private Branch Exchange (PBX) executive/administrator solutions, in which the administrator is able to barge in on executive calls.

The user can also specify whether a warning tone should be provided when a shared location barges in on an existing call. This warning tone is useful for privacy to notify the executive that another party is now listening on the call.

If the warning tone is enabled, a warning tone is provided to all parties connected to the bridge when a location barges in, and is optionally repeated periodically while the bridge is active.

A user at a Shared Call Appearance location can barge in on an active call by selecting the call and pressing a feature key (for example, Barge-in) on a SIP phone.

3.81.8.1 Barge-in versus Hold/Retrieve

The system distinguishes between retrieve and barge-in requests based on the call appearance state, that is, *held* or *active*. If two users attempt to retrieve a call appearance at the same time, then the first request is processed as a retrieve request and the second is processed as a barge-in request. Similarly, if a user attempts to barge in to a call appearance at the same time the other user holds the call appearance, then the barge-in request is processed as a retrieve request if it is processed immediately after the hold request.

3.81.9 Configuration

The group administrator configures the Shared Call Appearance service for the user.

There are eight different Shared Call Appearance services: SCA, SCA 5, SCA 10, SCA 15, SCA 20, SCA 25, SCA 30, and SCA 35. The service is chosen based on the maximum number of locations required for a given user. For example, if a user is assigned SCA 10, up to ten individual secondary locations can be defined for that user.

The administrator configures Shared Call Appearance locations and specifies whether to allow the termination and origination of calls for each location.

The configuration of SIP locations allows different line appearances on the phone to be assigned to different users. This allows applications such as an assistant/executive scenario and key system emulation.

The administrator controls the ability of users to enable or disable a secondary location. By default, users are allowed to enable or disable secondary locations.

Users can change Shared Call Appearance settings using the web portal, but they cannot add or remove locations. The following options are configurable by the user:

- Enable or disable alerting all appearances for Click To Dial calls.

- Enable or disable alerting all appearances for Group Paging calls.

- Allow or disallow call retrieval from another location.

- If the user has the Multiple Call Arrangement service assigned, enable or disable Multiple Call Arrangement.

- Allow or disallow bridging between locations.

- Enable or disable Call Park notification.

- Configure the bridge warning tone.

- Enable or disable secondary locations. This can be also done by using feature access codes.

3.81.9.1 Configuration of Multiple Call Arrangement with Shared Call Appearance

The Multiple Call Arrangement service is a user service and must be assigned to users.

It is configured through the Shared Call Appearance service profile. If the user has the Multiple Call Arrangement service but does not have the Shared Call Appearance service, then Multiple Call Arrangement is not configurable.

3.81.9.2 Configuration of Simultaneous Primary and Alternate Location Calls

System administrators define system-wide default settings for this capability. Enterprise and service provider administrators configure the capability for individual users.

The following can be configured:

- Allowing or disallowing simultaneous calls from primary and one alternate location

- Allowing or disallowing the use an alternate location to retrieve a current call on the primary location

- Allowing or disallowing the use an alternate location to barge in on a current call on the primary location

3.82 Silent Alerting

Silent Alerting allows the user to suppress ringing when incoming calls are received.

3.82.1 Description

The Silent Alerting service allows the user to suppress ringing on the following locations:

- Primary
- Shared Call Appearance
- Flexible Seating Guest
- Hoteling Guest
- Video Add-On

The user cannot select the devices on which to suppress ringing; they can only enable or disable the service. Silent Alerting only works if the device supports the proprietary Alert-Info header content. If the device does not support it, then the usual alerting is heard.

3.82.2 Configuration

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

3.83 Simultaneous Ringing Personal

The Simultaneous Ringing Personal service enables a user to have multiple destinations ring simultaneously when calls are received on their phone number. The first destination to answer the call is connected. This service is helpful, for example, when the user is not at their desk, and the user would like their cell phone to ring when they receive a call.

3.83.1 Description

This service is a "find-me" service that alerts multiple terminating locations simultaneously. A user can have up to ten secondary terminating locations (for example, cell phone, home phone, and MSN Messenger call client).

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

In addition, the user can specify selective criteria under which the service should ring secondary locations.

The criteria are based on the incoming caller identity, ranges of digits, time of day, and day of the week, and are combined into criteria entries, for example, an incoming call from this number, within business hours, and during the workweek. If required, multiple entries can be defined and used to screen calls. For more information, see section *6.18 Selective Criteria*.

If the call does not match any criteria entries, the call continues as if this service was not turned on. If, however, the service is active and there are no active criteria entries, the service rings the secondary locations for all calls.

Calls to secondary locations are subject to the services that apply to these locations. For instance, a call to a busy cell phone may be forwarded to voice mail, thus resulting in the other legs being released. If all call legs are busy, busy processing is applied to the call.

> **WARNING**: If a secondary location, such as a cell phone, has voice messaging that picks up before the office voice messaging picks up, messages could be left on the user's cell phone messaging system instead of the office messaging system.

The user can choose not to alert the secondary locations when the primary location is active on a call. The user can also turn off the service when they are at their desk.

The user can enable answer confirmation for each location, which allows Simultaneous Ringing Personal to prompt the called party to enter a digit to confirm the acceptance of the call. By alerting all destinations until a confirmation digit is received, it ensures that all of the called party's numbers are attempted before the call is rolled over to a voice mailbox.

3.83.2 Configuration

The user can configure the following:

- Activate or deactivate the service.

- Enter up to ten secondary phone numbers and specify for each location whether answer confirmation is required.

> **NOTE**: When the user enters a 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.

- Select whether secondary locations should be alerted while the primary location is active on a call.

- Define and activate/deactivate selective criteria entries.

3.84 Speed Dial 8

The Speed Dial 8 service allows a user to assign single-digit codes to frequently dialed or hard-to-remember long strings of digits. The user can then use these codes instead of the full numbers to place calls.

3.84.1 Description

A user can associate a single-digit code with a string of digits. This single-digit code is referred to as a speed code. The string can be between two through 30 digits long, and can contain numbers, and the * and # characters. This allows a user to program a speed code that maps to a feature access code or multiple chained feature access codes. The user can also associate a speed code with a SIP-URI, for example joe.blo@21.12.1.213.

> **NOTE**: Emergency, repair, and directory assistance numbers cannot be associated with a speed code. When a user tries to associate a speed code with an emergency or repair phone number, the Application Server prevents this action and returns an error.

Once a speed code is defined, the user can dial that single digit and wait for an inter-digit timer to expire or terminate the code with the # character. Once the speed code is collected, it is replaced by its associated digit string and the call is originated as usual with these digits.

3.84.2 Configuration

The user can use two methods to program speed codes. This can be through the CommPilot Personal web portal or by dialing a feature access code.

During programming, the system does not validate the digits. It is the user's responsibility to ensure that the address associated with a speed code is valid. When the associated address is invalid, the user is provided with the applicable treatment.

3.84.2.1 CommPilot Web Portal

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

3.84.2.2 Feature Access Code

The user can program Speed Dial 8 speed codes using a feature access code from their device.

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

Any changes to the user's Speed Dial 8 speed codes using a phone are reflected on the user's *Speed Dial 8* configuration page on the web portal.

3.85 Speed Dial 100

The Speed Dial 100 service allows a user to place calls using a directory of up to 100 frequently called numbers. The user dials the associated two-digit speed code, preceded by a configurable prefix.

3.85.1 Description

A user can associate a two-digit code with a string of digits. The string can be between two through 30 digits long, and can contain numbers, and the * and # characters. This allows the user to program a speed code that maps to a feature access code or multiple chained feature access codes. The user can also associate a speed code with a SIP-URI, for example joe.blo@21.12.1.213.

> **NOTE**: Emergency, repair, and directory assistance numbers cannot be associated with a speed code. When a user tries to associate a speed code with an emergency or repair phone number, the Application Server prevents this action and returns an error.

To make a call, the user enters the Speed Dial 100 prefix and a two-digit code. The user can then wait for the inter-digit timer to expire or terminate the code with the # character. Once the code is collected, it is replaced by the associated string and the call is originated as usual.

However, there are several situations in which other numbers can interfere with Speed Dial 100 dialing:

- Collision with a feature access code
- Collision with an emergency number or repair number
- Collision with an extension number

3.85.1.1 Collision with Feature Access Code

This scenario occurs when a speed code uses the same prefix and digits as an existing feature access code. For example, if the user assigns a phone number to a speed code for 76 and the Speed Dial 100 prefix is set to #, this may collide with the Call Forwarding Selective Activation feature access code because it is set to #76 by default. In such a case, the feature access code takes precedence and the speed code cannot be dialed.

Note that the default speed code prefix of the # character causes conflicts with the default feature access codes for several other Cisco BroadWorks services, for example, Call Waiting Persistent Deactivation (#43). To eliminate such conflicts, consider changing the prefix for the Speed Dial 100 service to another character or to ##.

3.85.1.2 Collision with Emergency Number or Repair Number

This situation occurs when a speed code plus its prefix is the same as an emergency or repair number. For example, if the Speed Dial 100 prefix is set to "9" and a speed code 11 is associated with 2403645239, this may collide with 911, when it is defined as an emergency or repair number. A user who tries to call 2403645239 using a Speed Dial 100 speed code would dial 9 (Speed Dial 100 prefix) and then 11 (the associated speed code), and would end up calling the emergency number 911 instead of 2403645239 since emergency numbers take precedence over speed codes.

3.85.1.3 Collision with Extension Number

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

Example:

A group has its extension dialing set to four digits and the Speed Dial 100 prefix is set to "52". The user assigns speed dial code 29 to a phone number and another user in this group has its extension number set to "5229". If the first user dials 5229, this would cause the extension number and not the speed code to be dialed. In this case, the extension takes precedence over the speed code.

3.85.2 Configuration

The administrator can configure the Speed Dial 100 prefix via the web portal, and the code can be one or two digits selected from the following: [0-9, A-D, *, #]. By default, the Speed Dial 100 prefix is the # character.

> **NOTE**: In IMS mode, the Speed Dial 100 prefix can be empty.

The user can use two methods to program speed codes. This can be done through the CommPilot web portal or through a phone using feature access codes.

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

3.85.2.1 CommPilot Web Portal

To configure a Speed Dial 100 speed code, the user selects the code and then enters the phone number/SIP-URI and description to associate with the code.

3.85.2.2 Feature Access Code

The user can program Speed Dial 100 speed codes using a feature access code from their device.

To program a speed code, the user dials *75 (default) from their device. The user is played the recall dial tone (three quick beeps and then a regular dial tone). The user then dials the two-digit speed code, followed by the digits to be associated with it and the terminating digit (#). The system plays a confirmation announcement and the user hangs up.

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

*75 [recall dial tone] 23 0112511792402# [confirmation announcement]

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

3.86 Three-Way Calling

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

3.86.1 Description

Users can invoke the service from an analog or IP phone.

3.86.1.1 Analog Phone

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

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

3.86.1.2 IP Phone

In addition to the conferencing functionality provided by Cisco BroadWorks, IP phones often provide embedded conferencing capabilities on the phone itself. Conferencing capabilities of IP phones are specific to each phone and vendor.

> **NOTE**: When a conference call is originated from an intelligent device, such as Polycom, the conference stays alive until the originator hangs up. This allows the originator to invite new participants to that call even if all the invited participants have left the conference.

3.86.2 Configuration

This service has no configuration.

3.87 Two-Stage Dialing

The Two-Stage Dialing service allows a user to integrate a cell phone or Public Switched Telephone Network (PSTN) landline with their Cisco BroadWorks services. It allows the user to dial their Cisco BroadWorks phone number followed by additional digits, and then use the additional digits for call origination attempts as if the user called from their Cisco BroadWorks phone.

3.87.1 Description

By associating the cell phone or PSTN landline with the user's primary device or alternate location, the system allows the user to originate calls from their cell phone or PSTN landline to their Cisco BroadWorks phone number. They can be prompted for additional digits, allowing them to leverage enterprise dialing and other Cisco BroadWorks services.

The service is triggered when a user originates a call from one of their configured locations, and the dialed digits match the user's Cisco BroadWorks phone number, an alternate phone number, the extension, an alternate extension, or a SIP alias.

Note that this service applies to all locations associated with the user via the Shared Call Appearance service.

The system answers the call and provides a special dial tone to the user. The user is optionally prompted based on the configuration and localization of the Two-Stage Dialing announcement. The user enters the destination digits. The digits are collected based on the public dialing plan and digit map corresponding to the user's group. The collected digits replace the dialed digits, and they are used to complete the call. An indication that the service was triggered is captured in a call detail record.

> **NOTE**: While the Media Server is collecting digits for the destination number dialed by the user during Two-Stage Dialing, the user cannot enter an E.164 number as the Media Server does not support the "+" sign for digit collection. The number used during Two-Stage Dialing should not be an E.164 number.

For a PSTN landline origination, the user's home phone is configured as a Shared Call Appearance on a system device that corresponds to the media gateway controller connected to the PSTN.

3.87.2 Configuration

The following configuration is required on Cisco BroadWorks to allow the user to use Two-Stage Dialing from a mobile phone or home phone number:

■ A system-wide device is created, which corresponds to the media gateway controller connected to the PSTN.

■ The system-wide device is assigned as the user's primary device. The line/port or identity of the primary device is set to the user's mobile phone number or home phone number.

-or-

The Shared Call Appearance (SCA) service is assigned to the user and the system-wide device is assigned as a secondary appearance. The line/port or identity of the secondary device is set to the user's mobile phone number or home phone number.

The user can turn Two-Stage Dialing on or off and configure it to use their primary phone number to activate the service.

3.87.2.1 Two-Stage Dialing Prompt

By default, the Two-Stage Dialing prompt file simply consists of silence. This feature was also designed for interworking with third-party dialer applications. In such cases, the silent mode is preferable from a user perspective and the silent mode is selected as the default.

The prompt can be localized so the behavior can be changed on a system-wide basis, similar to any other announcement. Upon upgrade, the customer must carry over that file manually.

Note that the BroadWorks Anywhere portal serves a similar purpose to Two-Stage Dialing. With BroadWorks Anywhere, the announcement is configurable on a per-portal basis.

3.88 UC-One Communicator

Part of the UC-One solution, UC-One Communicator is a client application for Unified Communications available for the desktop, tablet, and mobile environments.

3.88.1 Description

UC-One Communicator features include the following:

- High definition audio and video calling

- Instant messaging with both enterprise and federated contacts

- Rich presence capabilities based on many inputs, such as telephony, Microsoft Exchange, or user-selected status

- Desktop sharing

- My Room, an always-on collaboration space, enabling a cohesive experience for chat, audio, and/or video conferencing, and desktop sharing

- One-on-one or group-based communications

- Cisco BroadWorks call settings, for example, Call Forwarding and Do Not Disturb

- Voice mail support

- Branding with service provider's colors, themes, and naming

UC-One Communicator uses Cisco BroadWorks and BroadCloud services and resources to provide its functionality.

For a detailed description of the UC-One solution, see the *UC-One Connect Solution Guide*.

3.88.2 Configuration

UC-One Communicator is fully configurable in real-time from a centralized Device Management interface.

For information on the required setup and configuration of the UC-One Communicator, see the *UC-One Connect Solution Guide* and UC-One Communicator documentation.

3.89 Video Add-On

The Video Add-On service enables the use of video media in conjunction with regular audio media. If a user's primary device does not support video, this service can be used to configure a video-capable device to deliver the video portion of their call.

3.89.1 Description

This service allows an additional video-capable device to be configured for a user. It is especially useful when the user's primary device does not have multimedia capability. When the user receives a multimedia call, the system "splits" the call, directing the audio portions of the call to the primary device and the video portions to the video add-on device. All other services continue to operate without change on the audio portions of the call.
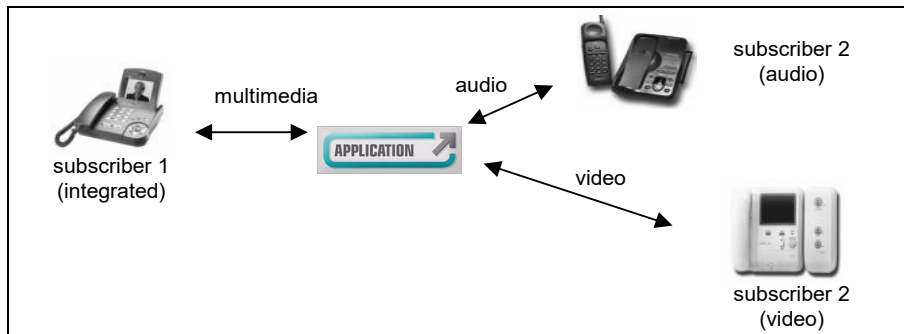
Figure 2 Video Add-On

If a user's primary device supports video, but the Video Add-On service is enabled, video is sent to the video add-on device, and not to the user's primary device. In addition, if the user's video add-on device is not reachable (it is not registered or it is not configured to be reached via an IP address/host name) but the service is enabled, then the call proceeds as an audio-only call regardless whether the primary device supports video.

When the service is activated, it automatically applies to all calls to and from the user. If the user receives an incoming video-only call, the system rejects this invitation since video-only calls are not supported by Cisco BroadWorks at this time.

3.89.1.1 Video Origination Time-Out

When the user makes or receives a call involving video, the video client is automatically alerted and a configurable (per-user) timer is started. This video invitation timer specifies the amount of time that the call can be delayed waiting for the video device to answer.

If the timer expires before the user's video device answers, then the video session is canceled and the call completes as an audio-only call.

3.89.1.2 Click To Dial Originations

If the user dials a number using a Click To Dial-enabled device, the system first sends an invitation to the user's primary device. An invitation to the user's video device is not sent until the user has picked up the handset on the audio device.

3.89.2 Configuration

The administrator configures the video add-on device for the user. The user can enable or disable the service and specify the maximum time to wait for the video device to answer.

3.90 Yahoo! Messenger Online Status

This feature allows a user's Yahoo! Messenger online status to be shown on the *Group Directory* page.

3.90.1 Description

This feature allows a user to enter their Yahoo! Messenger user ID on their *Profile* page. It also allows the administrator and the user to view the online status of users within that group in the group directory (or enterprise directory, in the enterprise model), provided that the users have configured their Yahoo ID.

3.90.2 Configuration

This service does not need to be assigned to the user. The user configures their Yahoo! Messenger user ID.

3.91 Zone Calling Restrictions

This service provides the ability to assign home zones to users, based on their actual location, and to apply calling restrictions on services such as forwarding, transferring, or bridging a Cisco BroadWorks user with a Public Switched Telephone Network (PSTN) party from another zone.

3.91.1 Description

Cisco BroadWorks users can be assigned a home zone, selected from a predefined list of zones that identify the area where the user is located.

The following enterprise-level policies can be independently enabled or disabled:

- Zone Calling Restrictions
- Origination Roaming Restrictions
- Emergency Origination Roaming Restrictions
- Termination Roaming Restrictions

When the Zone Calling Restrictions policy is enabled, users who have been assigned the Zone Calling Restrictions service are subject to the following rules:

- Redirections are limited to the same zone destinations if a PSTN party is involved in the call.
- Conferences are limited to the same zone destinations if a PSTN participant is involved in the conference.
- Roaming users are prevented from originating or terminating off-net (PSTN) calls, including redirected on-net calls.
- Users in an enterprise can always call each other freely regardless of zones.

> **NOTE**: PSTN in this context denotes any destination from outside the user's enterprise or group (in the service provider model).

When the Origination Roaming Restrictions policy is enabled, users with the Zone Calling Restrictions service assigned are prevented from originating non-emergency calls from any roaming device.

When the Emergency Origination Roaming Restrictions policy is enabled, users with the Zone Calling Restrictions service assigned are prevented from originating emergency calls from any roaming device.

When the Termination Roaming Restrictions policy is enabled, users with the Zone Calling Restrictions service assigned are prevented from receiving calls on any roaming device.

The Zone Calling Restrictions service can also be assigned to virtual subscribers.

3.91.2 Configuration

The system administrator defines a system-wide list of zones, using the CLI.

For each zone, system providers can provision a list of IP address ranges that specify which devices (according to their IP addresses) are physically located in a given zone. These are used to determine whether a user is roaming. Each zone is also provisioned with a physical location.

4 Group Services

This section describes the services offered to Cisco BroadWorks groups. It includes services that apply to all users in the group, multiuser services, and services configured as virtual users.

Group services are typically authorized to specific groups by the service provider or enterprise administrator and assigned, managed, and configured by the group administrator via the web portal. This information is not explicitly mentioned in each section; however, departures from this model are specified.

4.1 Account Codes

The Account Codes service is a multiuser service that enables the tracking of calls made outside of the group by prompting selected users for an account code.

4.1.1 Description

This service allows users to assign certain calls to specified accounts, for tracking purposes. The entered codes are not validated (see the Authorization Codes service), so calls are not blocked if an invalid code is entered. Note that groups cannot have this service and the Authorization Codes service enabled at the same time.

Two account code-dialing methods are offered, which can be assigned concurrently to different users of a group: mandatory and feature access code-based (optional).

4.1.1.1 Mandatory Usage

Users who are required to enter an account code are prompted for an account code every time they make a call outside of the group. When prompted to enter a code, the user dials the applicable digit string, after which the call resumes as usual. The code is captured in the associated accounting information generated for that call.

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

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

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

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

4.1.1.2 Feature Access Code-Based Usage

Users for whom the use of access codes is optional can:

- Make a call as usual, without entering an account code.
- Dial the Per Call Account Code feature access code (for example, *71) before making a call. In this case, the user is prompted for an account code, and the user dials the code, receives confirmation, and then proceeds with the call as usual. The sequence is as follows:

[Off-hook] [FAC] [prompt] [code] [confirmation] [dial tone] [call]

■ Flash the switch hook during the call and enter the Per Call Account Code feature access code (for example, *71). In this case, the user is prompted for an account code; the user dials the account code, and then the user is reconnected to the call. The sequence is as follows:

[Call] [flash] [FAC] [prompt] [code] [confirmation] [call]

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

Users for whom the use of access codes is optional cannot make a blind transfer by entering [FAC] [code] [destination number].

4.1.2 Configuration

The group administrator configures the following:

■ Activates or deactivates the service

■ Selects the length of the account code

■ Selects whether the service should apply only to long distance calls

■ Selects which users of the group are assigned the service

■ Selects the method to be used for each user

■ Configures account codes

Users can view the configured account codes but they cannot modify them.

4.2 Authorization Codes

The Authorization Codes service is a multiuser service that performs an authorization of calls made outside of the group by prompting selected users for an authorization code.

4.2.1 Description

This service allows the group administrator to select specific users who must enter a valid authorization code when making a call to a party outside of the group. Calls are not connected unless a valid code is entered. Note that groups cannot have this service and the Account Codes service enabled at the same time.

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

When a user dials a number for which an authorization code is required, the system prompts the user for a valid code. The user then dials an authorization code followed by the (#), or waits for the inter-digit time-out. If the code entered is not valid, the user is provided with another attempt at entering a valid code. If the second attempt is also unsuccessful, the user is provided with the applicable treatment. If a valid code is entered, the call is allowed to proceed as usual and the authorization code entered by the user is captured in the accounting information generated for that call.

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

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

> **NOTE**: The Authorization Code service applies to all applicable calls made from a device assigned to a user who is subject to the service.

4.2.2 Configuration

The group administrator configures the following:

- Activates or deactivate the service
- Selects the length of the authorization codes
- Selects whether non-toll calls are subject to an authorization code
- Configures valid authorization codes
- Selects which users in the group are required to use authorization codes

4.3 Auto Attendant

The Auto Attendant is a powerful and flexible tool for fielding inbound calls and delivering them to the intended destination through interactions with the caller. The Auto Attendant is an integral part of the Cisco BroadWorks product offering and does not require an external third-party system. If is offered in two editions.

4.3.1 Description

The Auto Attendant serves as an automated receptionist that answers the phone and provides a personalized message to callers with options for connecting to the operator, dialing by name or extension, leaving a message, listening to an announcement, or connecting to configurable destinations (for example, 1 = Marketing, 2 = Sales, and so on).

Auto Attendant is offered in two editions allowing businesses to choose the solution that best suits their needs: Auto Attendant – Basic and Auto Attendant – Standard.

- Auto Attendant – Basic service provides support for direct transfer to voice mail and listening to custom announcements, in addition to the automated receptionist capabilities described earlier.

- Auto Attendant – Standard service provides support for submenus and holiday schedules in addition to the capabilities of the Auto Attendant – Basic service.

To reach the Auto Attendant, the caller dials an associated phone number or extension. Once connected, the caller is played a greeting that provides a menu of options to complete call routing.

The menu, which is configured by a group administrator, may include:

- **One-Key Dialing** – The caller presses a predefined DTMF key to reach a particular phone number or extension within the group.

- **Operator Dialing** – The caller presses a predefined DTMF key to reach an operator.

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

  The Auto Attendant may be configured to allow name dialing with the first name entered before the last name, in addition to dialing by the last name followed by the first name.

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

- **Immediate Extension Dialing** – The caller dials the desired extension right away, without having to first navigate to the Extension Dialing menu.

- **Voice Messaging** – The caller dials the desired extension and the caller is immediately forwarded to the extension's voice mailbox.

- **Listening to Announcement** – The caller presses a predefined DTMF key to hear a custom announcement, such as company's business hours, provided services, special events, and so on.

The Auto Attendant can be configured to provide a different service during business hours and outside business hours including:

- **Enhanced Business Hours Support** – A group administrator can associate a time schedule with an Auto Attendant. Outside business hours, callers hear the after-hours greeting and dialing menu, configurable independently of the business hour greeting and dialing menu. The administrator has the option to provide 24/7 service.

- **Holiday Schedule** – A group administrator can associate a holiday schedule with an Auto Attendant. On a scheduled holiday, callers hear the after-hours greeting and dialing menu (Auto Attendant and Auto Attendant –Basic) or the holiday greeting and dialing menu (Auto Attendant – Standard).

- **Transfer to Submenu** (Auto Attendant – Standard) – A group administrator can associate a submenu with a DTMF key. The caller presses the predefined DTMF key and the user is transferred to the submenu. This option is used to build multilevel Interactive Voice Response (IVR) menus as shown in the following example.
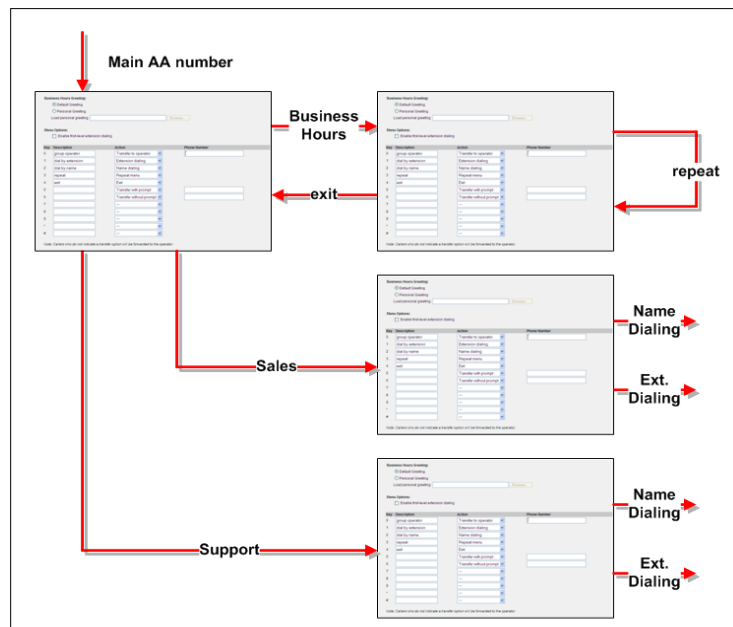
Figure 3 Multilevel IVR Menus

For Auto Attendant and Auto Attendant – Basic, this can be obtained by defining multiple Auto Attendants and integrating them into a multilevel Auto Attendant using One-Key dialing. For example, an enterprise's main Auto Attendant can be configured to route to the Auto Attendant of a particular department or location.

4.3.1.1   Automatic Access to Users in Group

The Auto Attendant has access to user information for name and extension dialing functions. The moves, additions, and changes for users in a group are automatically available to the Auto Attendant.

4.3.1.2   Dedicated Name for Name Dialing

The administrator has the ability to configure the user's first and last name dedicated to name dialing. The Auto Attendant then searches using these names for name dialing.

4.3.1.3 Multisite Support

The Auto Attendant uses the multi-location enterprise capabilities of the Cisco BroadWorks platform to support geographically distributed groups transparently.

4.3.1.4 Support for Users Without DID

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

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

4.3.1.5 Video Support

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

Video support is automatically provided with Auto Attendant – Standard. To provide video support in Auto Attendant or Auto Attendant – Basic, the corresponding video service (Auto Attendant – Video and Auto Attendant – Basic Video respectively) needs to be assigned to the group.

4.3.2 Configuration

The group administrator may create one or more Auto Attendants for the group. The Auto Attendant configuration includes the following:

- **Services** – The group administrator can assign services such as Call Forwarding or Do Not Disturb to an Auto Attendant.

- **Cisco BroadWorks voice portal greeting** – The group administrator can record new greeting menus through the voice portal phone interface. This automatically provisions the newly recorded greeting as the active greeting for the Auto Attendant.

- **Video support** – To enable video support for the Auto Attendant or Auto Attendant – Basic, the group must have the Auto Attendant – Video or Auto Attendant – Basic Video service assigned.

- **Scope of extension and name dialing** – The administrator can restrict the scope of extension dialing and name dialing to a department.

- **Name order in name dialing** – The administrator can configure Auto Attendant to allow name dialing with the first name entered before the last name, in addition to the last name entered before the first name.

- **Business Hours** – The administrator can select a business hour schedule from a previously defined list of time schedules.

- **Holiday Schedule** – The administrator can select a holiday schedule from the previously defined list of holiday schedules.

- The administrator can configure a different greeting and menu options for business hours, after hours, and holidays (Auto Attendant – Standard).

- The administrator can configure the time-out for the first digit collection when a caller is navigating through the top-level Auto Attendant menus.

The following options are provided on the Auto Attendant *Business Hours Menu*, *After Hours Menu, Holidays Menu,* and submenus:

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

■ **First-level extension dialing** – The group administrator can enable immediate extension dialing that allows a caller to dial an extension right after the greeting.

■ **Default menu options** – The group administrator can assign keys to the default menu options of the Auto Attendant.

■ **Customized menu options** – The group administrator can create customized menu options by associating keys with specific phone numbers, custom announcements, and submenus.

■ **Customized actions** – The group administrator can assign specific actions to the keys entered by the user, such as transferring the call to an operator, phone number, voice mailbox, or submenu; or name or extension dialing; or listening to a custom announcement.

NOTE: Some options are only available in a specific mode of Auto Attendant edition as described in the previous section.

The group administrator can also configure Cisco BroadWorks users with a first and last name dedicated to name dialing.

4.4 BroadWorks Anywhere Portal

The BroadWorks Anywhere portal is a virtual user service that handles incoming calls from the BroadWorks Anywhere locations and prompts users for the destination address.

4.4.1 Description

The BroadWorks Anywhere portal works in conjunction with the BroadWorks Anywhere user service. It performs the following operations for the user:

- Handles incoming calls from the BroadWorks Anywhere locations
- Prompts users for the destination address

The BroadWorks Anywhere portal serves a purpose similar to the Two-Stage Dialing service. Once the destination digits are collected, the call proceeds as if the call originated from the user's desk phone.

4.4.2 Configuration

The group administrator defines the BroadWorks Anywhere portal service instances (virtual users). A group may have one or more BroadWorks Anywhere portals, each with its own characteristics.

A BroadWorks Anywhere portal virtual user is configured similarly to other virtual users. Additional options available specifically for BroadWorks Anywhere portal include configuring the prompt to use and the scope of the portal.

Depending on the type of BroadWorks Anywhere deployment, an administrator may need to provide multiple access points to BroadWorks Anywhere users. For example, if the service is meant to be available in different countries, then the administrator may want to provide an access point in each country to increase the likelihood that the calling address is available to the BroadWorks Anywhere portal.

4.5 Call Center

Call Center is a virtual user service that allows business agents to receive incoming calls from a central phone number. Using this service, a business can establish technical assistance lines, customer support numbers, or order-taking centers. Multiple call centers can be supported for each business. Incoming calls to a call center are presented to agents according to the selected Call Distribution policies. Call centers can have multiple phone numbers with different priorities, allowing a finer prioritization of calls.

Cisco BroadWorks provides a three-tiered call center offering allowing businesses to choose the solution that best suits their needs.

4.5.1 Description

This service builds on the basic Hunt Group service to provide a complete, business-ready application. As a result, call centers inherit the characteristics of the Hunt Group service. They are provided with sophisticated call-handling features, such as Queuing, Music On Hold, and so on. For a complete description of Hunt Group service features and characteristics, see section *4.18 Hunt Group.*

The Directory Number Hunting service can also be assigned to a call center, which allows a caller to reach a call center by calling the number of one of the call center agents. For more information on this service, see section *4.18.1.4 Directory Number Hunting.*

Call Center service functionality can be combined with other Cisco BroadWorks Call Completion services to ensure that all incoming calls are serviced expeditiously under any network condition and at any time.

Group administrators assign agents and supervisors to call centers. An agent needs a call center license at the appropriate level before they can be assigned to a call center. A supervisor can be any member of the group or enterprise and can be assigned to more than one call center. A supervisor then selects agents to be supervised.

Provisioning interfaces are available to support third-party call center supervisor and agent clients.

4.5.1.1 Call Center Offerings

Call Center provides three levels of functionality allowing businesses to choose the solution that best suits their needs:

- **Call Center – Basic** – This option provides basic queuing capabilities distributing calls to devices and people. This is an inexpensive option for multiple receptionists and other call center applications that require basic queuing capabilities.

- **Call Center – Standard** – In addition to basic queuing capabilities provided by the Basic solution, this option also provides such functions as agent states, supervising, and reporting. It is a balanced option taking into consideration features and price.

- **Call Center – Premium** – This option is the most feature-rich Call Center solution provided by Cisco BroadWorks. In addition to the capabilities provided by the standard offering, it provides features such as Night Service, Stranded Calls, and Bounced Calls Routing policies. It is intended to meet the call center needs of most businesses.

The following table summarizes the functionality of the different call center offerings provided by Cisco BroadWorks.

Feature	Basic	Standard	Premium
Call Processing			
Maximum Calls in Queue	25	50	525
Call Distribution to Agents	X	X	X
Call Escalation to Supervisor		X	X
Emergency Escalation	X	X	X
Multiple Queue Priority Routing			X
Uniform Call Distribution Priority Scope	X	X	X
Call Queuing	X	X	X
Distinctive Ringing	X	X	X
Distinctive Ringing for Forced Delivery of Calls			X
Call Center Call Information	X	X	X
DNIS Support			X
Call Statistics Reset			X
Overflow	X	X	X
Bounced Calls	X	X	X
Stranded Calls		X	X
Forced Forwarding			X
Night Service			X
Holiday Service			X
Forced Delivery of Calls			X
Allow Call Waiting on Agents	X	X	X
Allow Calls to Agents in Wrap Up State		X	X
Set Maximum ACD Wrap-up Timer		X	X

Feature	Basic	Standard	Premium
Call Treatment and Announcements			
Basic Announcements	X	X	X
Comfort-Bypass Messaging			X
Queue Escape	X	X	X
Call Treatment Chaining	X	X	X
Agent and Supervisor Support			
Call Center Support for Agent and Supervisor Clients		X	X
Third-Party Call Center Clients		X	X
Geographic Distribution	X	X	X
Agent ACD State		X	X
Allow Agents to Join Call Centers	X	X	X
Agent Sign-in/Sign-out		X	X
OSS Access	X	X	X
Agent Guard Timer Settings	X	X	X
Standard Agent Availability Settings and Agent Unavailable Codes		X	X
Disposition Codes			X
Agents Availability on Personal Calls			X
Agent Hold and Retrieve		X	X
Call Center Outgoing Calls			X
Agent Whisper Message			X
Supervisor Support		X	X
Silent Monitoring (only agents with Call Center – Premium assigned can be monitored)		X	X

Feature	Basic	Standard	Premium
Statistics and Reporting			
Call Center Status and Statistics		X	X
Call Center Reporting		X	X
Call Recording	X	X	X
Service Integration			
Voice Mail	X	X	X
Integration with Cisco BroadWorks User Services	X	X	X
Outlook Contact Integration	X	X	X

4.5.1.2   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 Cisco BroadWorks 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 section *4.18 Hunt Group*.
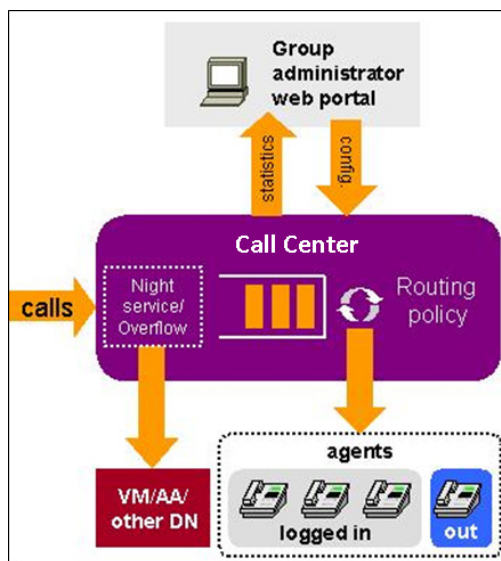
Figure 4 Call Distribution Policies

- **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 the system 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** – If all agents in the call center group log out, then 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 should 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 section *4.5.3 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 Cisco BroadWorks platform. The third-party Call Center clients use the Cisco BroadWorks Open Client Interface (OCI) to communicate with Cisco BroadWorks.

- **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 Cisco BroadWorks 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 they 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. In addition, only agents with the Call Center – Premium service assigned can be silently monitored.

- **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 section *4.5.2 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** – Most Cisco BroadWorks personal services 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.

4.5.2   Call Center Reporting

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 email 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.

4.5.3 Agent and Supervisor Clients

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

4.5.3.1 BroadWorks Agent

BroadWorks 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.

An Xtended Services Interface application ID is created for the BroadWorks Agent. Clients using the BroadWorks Agent as the application ID can log in only with the BroadWorks Agent service assigned. These clients have complete ability to use Xtended Services Interface 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 Call Center client supports this capability.

4.5.3.2 BroadWorks Supervisor

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

An Xtended Services Interface application ID is created for the BroadWorks Supervisor. Clients using BroadWorks Supervisor as the application ID can log in only with the BroadWorks Supervisor service assigned. These clients have complete ability to use Xtended Services Interface call control messages.

4.5.4 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

- Agent's settings and agent's unavailable codes

- Services

- Disposition codes that can be assigned to calls

- DNIS numbers

4.6 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.

4.6.1 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 1**: A call cannot be parked against a remote party.
>
> **NOTE 2**: There must be only one other active call when a call park is attempted. If there were 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.

Once a call is parked, the user can hang up or perform other telephone tasks.

Once the call is parked against an available user, the service updates the parking user's display with the extension where the call is parked and provides a visual indication at the extension where the call is parked that a call is parked against that extension.

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.)

4.6.1.1   Call Park Recall

A configurable recall timer is started when a user parks a call. If the timer expires before the parked call is retrieved, the system 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.

4.6.2 Configuration

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

The group administrator can also:

- Configure a default audio on hold for the group, which is played to all parked calls.

- 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.

- Define one or more call park groups.

4.7 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.

4.7.1 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. Users then flash the switch hook to toggle between the two parties.

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. For information, see section *3.46 Directed Call Pickup*.

4.7.2 Configuration

The group administrator defines call pickup groups through the CommPilot 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.

4.8 Calling Plan

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

4.8.1 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.

4.8.1.1 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.

4.8.1.2 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. (The system 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 user

    – 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 call types:

- **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 email 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.

4.8.1.3 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 Cisco BroadWorks.)

- **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.

4.8.1.4 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.

4.8.1.5 Interaction between ODP, OCP, and OPDP

The following outgoing calling plans are available on Cisco BroadWorks:

- 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, the system 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.

4.8.1.6 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 the following virtual users:

- Meet-Me Conferencing
- Route Point

4.8.2 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.

4.9 Configurable Feature Access Codes

Group administrators can select the FACs used to activate, deactivate, and program various Cisco BroadWorks services.

4.9.1 Description

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

A feature access code is defined as a string of two through five digits and special characters that are associated with a Cisco BroadWorks 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.

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

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

The system validates all the feature access codes to ensure that:

- There are no collisions between feature access codes.

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

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

When there is a collision between a feature access code and an extension or a speed dial code, the feature access code has precedence.

4.9.2 Configuration

This feature is automatically available with Cisco BroadWorks. By default, feature access codes configured by the service provider/enterprise administrator are used for the group.

4.10 Custom Ringback Group

The Custom Ringback Group service allows an audio ringback to be defined for the group. When a group member is called, the system allocates a media resource and plays a custom ringback file to the caller instead of the standard ringback tone.

4.10.1 Description

The group administrator can configure a custom ringback to be played to the users in the group.

In addition, the Custom Ringback Group – Video service is also available. It enhances the Custom Ringback Group service by allowing a video ringback to be configured for the group in addition to audio ringback.

Custom Ringback audio and video files loaded via the web interface are subject to a length and format validation. This applies to media files loaded as files, either from the web interface or from the OCI/OSS interfaces. Although this validation is not done for ringback files specified as URLs, it is recommended to use ringback media files of the allowed length.

Note that users with the Custom Ringback User service assigned can override the group ringback settings.

4.10.2 Configuration

The Custom Ringback Group – Video service requires the Custom Ringback Group service to be assigned to the group. It cannot be used on its own.

The group administrator downloads the audio and/or video ringback files and enables or disables the service.

4.11 Department Administrative Layer

This feature provides a department administrative layer to help the management of large or geographically distributed groups.

4.11.1 Department Administrators

Similar to groups, departments can be assigned an administrator who shares many of the group administrator's privileges for the users and service instances assigned to the department. As a result, the group administrator can delegate most of the day-to-day service management of the department to a department administrator.

Specifically, department administrators have the following management capabilities:

- Add, modify, and delete users in their department.

- Modify service instances, such as Auto Attendants, Call Centers, and Meet-Me conference bridges that belong to the department.

- View group directory and modify group common phone list.

The group administrator defines the administrative scope of the department administrators by assigning users and virtual users (service instances) to specific departments. The department administrators can manage users and service instances that belong to their department the same way the group administrator can.

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

4.11.2 Department Provisioning

Instances of some virtual user services, such as Auto Attendant and Call Center, can be assigned to a department. When a service instance is assigned to a department, the department administrator can manage it.

In particular, assigning Auto Attendant to a department allows the option to restrict the scope of name dialing to the users in the department. This is especially useful with large groups as it can limit the number of collisions between user names.

Furthermore, the CommPilot web portal allows the administrator to bulk-provision services with entire departments. As a result, when populating the agents in a call center, the administrator can select a whole department at once instead of selecting the users one by one. Services such as Call Center, Account and Authorization Codes, Call Capacity Management, and Call Pickup are enhanced with department bulk provisioning.

Finally, group administrator can assign phone numbers (DID) to departments. Any phone number assigned to a department shows up as such on the list of phone numbers when assigning new phone numbers to users or group services. The department phone number can be used as the calling line identity for users in the department.

4.11.3 Configuration

Group administrators create departments and only group administrators can assign services instances to a department. They can also assign users to departments. Department administrators can create users. Any user created by the department administrator is automatically assigned to the same department.

Once users and services are assigned to specific departments, they can be configured and managed by the department administrators. Department administrators can delete any user in their department; however, they cannot delete virtual users.

NOTE: The use of departments is optional. Department selection for users and services may be omitted, which leaves configuration and management at the group level.

4.12 Device Inventory

The Device Inventory feature provides the group administrator with an inventory of all integrated access devices (IAD), gateways, and IP phones for a group. In addition, this feature provides the capability to assign users directly to a device and/or a port on a device.

4.12.1 Description

This feature allows the group administrator to keep track of the devices (IADs, gateways, IP phones) used by the group. It also hides the technical details specific to each device and vendor so the group administrator can easily assign users to devices without an in-depth understanding of how the device works.

For instance, to activate User A, the group administrator plugs User A's phone into port N of device X, and then assigns User A to port N or device X through Cisco BroadWorks device inventory, without having to know the specific port numbering scheme used for that device and vendor.

The Device Inventory feature also allows the group administrator to drill down into a device by clicking on it on the inventory list to view and modify detailed information regarding the device and to see all users assigned to this device.

4.12.2 Configuration

This feature does not need to be authorized or assigned to groups.

The group administrator can add, delete, and modify the devices for the group.

When the administrators add new devices to their group, they are required to pick the device type from the list of available device types known to Cisco BroadWorks. The new device then becomes a new instance of a known device type, and the system can present a standardized configuration page to the administrator, which abstracts all the device specific details.

4.13 Emergency Zones

The Emergency Zones service allows a service provider to decide on a group basis whether emergency calls are allowed for a user when roaming outside of the group's home zone or location.

4.13.1 Description

The service provider can define a home zone or location for a group and deny SIP registrations, call originations, or emergency call originations based on the home zone.

When SIP registrations and call originations are not allowed and a user in the group tries to use their SIP phone or device from outside of the home zone, the device is not able to register or originate calls.

On a registration attempt by the SIP phone or device, the system responds with an error and triggers a system alarm.

On a call origination attempt, the user receives a treatment. This is a new treatment configurable at the system level.

When emergency calls from outside the home zone are not allowed, SIP users can make calls from outside of the home zone, but they cannot make emergency calls, thus preventing invalid locations being provided to emergency response teams based on the user's calling line ID.

If an emergency call is denied, then the user receives a treatment.

The service provider can also optionally configure an email address where detailed call information for emergency calls made from a user is sent. This email is sent whether the emergency call is permitted or denied, as long as an address is configured and the option of sending emails is enabled.

The email message 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).

4.13.2 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 email.

4.14 Extension Dialing

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

4.14.1 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 it 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 and from Click To Dial-enabled clients.

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 section *4.28 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.

4.14.2 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.

4.15 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, Cisco BroadWorks 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.

4.15.1 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.

4.15.1.1 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 Cisco BroadWorks users who are part of that Find-me/Follow-me group. Any Cisco BroadWorks 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.

4.15.2 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 CommPilot 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 Cisco BroadWorks 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.

4.16 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.

4.16.1 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 CLID number defined, the department phone number can be used instead of the group phone number as the calling line identity.

4.16.2 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.

4.17 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.

4.17.1 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.

4.17.1.1 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.

4.17.1.2 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 BroadWorks 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.

4.17.1.3 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.

4.17.1.4 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

4.17.2 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 CommPilot 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.

4.18 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.

4.18.1 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.

4.18.1.1 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.

Cisco BroadWorks 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.

4.18.1.2 User Services

As hunt groups are virtual users, they can be assigned some users services, for example, Anonymous Call Rejection, Directory Number Hunting (see section *4.18.1.4 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.

4.18.1.3 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 or the UC-One Communicator, if assigned.

- **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.

- **BroadWorks Anywhere** – The BroadWorks 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.

4.18.1.4 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 it cannot be added to a service pack.

4.18.1.5 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.

4.18.2 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 device.

- 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.

4.19 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.

4.19.1 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).

4.19.2 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 a FAC, provided 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.

4.20 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.

4.20.1 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 BroadWorks 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.

4.20.2 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.

4.21 Loudspeaker Paging

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

4.21.1 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.

4.21.2 Configuration

The loudspeaker paging system is configured similar to a regular Cisco BroadWorks 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.

4.22 Meet-Me Conferencing

Meet-Me Conferencing provides superior functionality over the Legacy Audio Conferencing application offered by Cisco BroadWorks. In addition to the many features provided by 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.

4.22.1 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 Cisco BroadWorks 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 Cisco BroadWorks 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 Cisco BroadWorks 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 *Cisco BroadWorks Meet-Me Conferencing Guide*.

4.22.2 Configuration

The system administrator configures general Meet-Me Conferencing settings 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.

4.23 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).

4.23.1 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.

4.23.1.1 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. The system 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.

4.23.2 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.

4.24 Outside Access Code

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

4.24.1 Description

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

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

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

4.24.2 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.

4.25 Resource Inventory Reporting

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

4.25.1 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 email to the specified address as an ASCII comma-separated value (CSV) attachment. The resources reported include:

- Phone numbers
- Devices
- Users and departments
- Services

4.25.2 Configuration

The group administrator provides the email 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

4.26 Route Point

The Route Point service allows a remote application (for example, an external call center) to queue and route calls.

4.26.1 Description

The Route Point service enhances Cisco BroadWorks by allowing an external system to integrate with Cisco BroadWorks easily to provide enhanced call center functionality.

The service provides functionality similar to that of the Call Center – Premium service; however, it does not perform call distribution and reporting functions. It performs the following functions on behalf of the external system:

- Queuing of calls

- Sending notifications to the external system about incoming calls

- Handling call management actions requested by the external system and applying them to specific calls in the queue

A route point is a call center queue that performs the same function; however, it allows an external system to distribute calls instead of making these decisions itself.

The following table summarizes the functionality offered by Route Point.

Function	Description
Call Queuing	Incoming calls are queued waiting for instructions from the external system unless some special treatment applies to them, such as overflow routing. The route point sends a message to the remote system that a new call has been added to the queue.
Handling of Remote System Requests	This functionality consists of applying actions on route point calls and resources according to instructions received from the remote application.
Agent and Supervisor Support	Agents are not assigned to route points. A user with Call Center – Premium service assigned can receive calls from route points. Any user can be a route point supervisor. The agents need to be assigned supervisors for call escalation and emergency calls.
Call Escalation and Emergency Calls	The ability to escalate and perform emergency calls via the customer premises equipment (CPE) is supported for route points using the configured supervisor – agent mapping.
Monitoring Calls	Users with Directed Call Pickup with Barge-in and Call Center Monitoring services can silently monitor route point or agent calls.
Agent Unavailable Settings	Agent Unavailable settings can be configured to change the agent's status automatically to *unavailable* in specific situations such as answering a personal call. In addition, agents may be required to specify the reason for their unavailability by entering an agent unavailable code.
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.
Call Distribution to Agents	When an inbound call is received and added to the queue, the route point does not automatically distribute the call to an agent. Instead, it waits for a request from the remote application and offers the call to the agent specified in the request.

Function	Description
Forced Delivery of Calls	The Route Point service supports a configuration option that allows calls to agents to be answered automatically and rendered over the device's speaker and microphone. The agent's device must support the Remote Control Talk event package.
Distinctive Ringing	The Route Point service offers the option to provide distinctive ringing to agents when the call is routed from the route point.
Blind Call Transfer	The route point blind transfer request is used by the remote application to transfer a call queued in the route point queue to a new destination. The new destination is identified in the request.
Processing Outgoing Calls	The Route Point service provides the ability for a remote application to originate a call from a route point to a target destination.
Releasing Call	Calls in the queue can be released by the remote application. The call to be released can be an inbound call or an outbound call.
Failing Calls	The remote application can explicitly request a route point to mark a specified call as failed and apply the Per-Call Failure policy to the call. The call must be an inbound call that has not been offered to an agent.
DNIS Support	The Dialed Number Identification Service (DNIS) provides association with multiple DNIS numbers, in addition to the primary number, with a route point. A calling party can dial a DNIS number associated with a route point instead of the primary number, to reach a route point. The remote application can implement a per-DNIS behavior for a route point. For instance, it can play different announcements and distribute calls to agents based on the DNIS on which the call was received. The DNIS number can be enabled for use by agents to make outgoing calls.
Agent Outgoing Calls	Agents assigned the Call Center – Premium service can make outgoing calls as a route point using route point primary or DNIS numbers.
Messaging	
Music/Video On Hold	Route point can be configured to play audio and video messages to callers in queue.
Treatment Messages	An external system specifies a treatment to play to the queued caller.
Agent Whisper Message	The agent whisper consists of playing an announcement to the agent while offering a call. The Agent Whisper URL is optionally provided by the remote application in the request for distributing the call to the agent.
Routing Policies	
Overflow	This policy allows the route point to processes calls that have been waiting in queue longer than the specified threshold.
Bounced Calls	This processes calls unanswered by agents. Calls can either be kept in queue or transferred to a configurable number.
Forced Forwarding	The Forced Forwarding policy allows calls to be temporarily diverted to a configured destination.
Night Service	This allows the route point to treat calls differently outside of service hours.
Holiday Service	Route points can be configured to treat calls differently during holidays.

Function	Description
External System Monitoring and Failover Policy	The Route Point service has the ability to monitor route point external systems and to route calls to an alternate destination in case of external system failure. A manual failover can be used to shut down an external system gracefully.
Per-Call Failure Policy	The system monitors the calls in the queue and if the remote application does not perform an action on a call for a configurable period of time, the Per-Call Failure policy is applied to the call. The policy can be enabled or disabled by the remote application.
Integration with Cisco BroadWorks User Services	A route point is a virtual user and can be assigned user services such as Call Forwarding services, Voice Mail services, Call Recording, Do Not Disturb, and so on.

4.26.2 Configuration

Route points are virtual users created at the group level. The group must have the Route Point service assigned and activated to be able to create route points. Each route point consumes one route point license.

The group administrator can configure the following:

- Basic information, such as name, calling line identity, department, and time zone

- Phone number, extension, and aliases

- Bandwidth and QoS settings

- Route point profile settings, such as queue length, and support for video announcements

- Call routing policies (such as, Overflow, Night Service, Bounced calls, or Stranded Calls) settings

- Call center announcements attributes specifying the source for each type of announcement

- Statistics and reporting settings, such as statistics source, statistics sampling frequency, reporting period, and destination address

- Agent attributes, such as maximum wrap-up timer, post-call ACD state, and forced delivery of calls

- Services assignment and settings

- Failover policy

- Disposition codes that can be assigned to calls

- DNIS numbers

At the system, enterprise, and group levels, defaults for agent availability can be configured. In addition, at the enterprise and group levels, agent unavailable codes and enterprise/group level disposition codes can be defined.

The system administrator needs to assign external systems to service providers and enterprises, which then need to be assigned to groups.

At the user level, group administrators need to assign supervisors to users who will be staffing route points, so that the agents can escalate calls.

4.27 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.

4.27.1 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 Cisco BroadWorks configuration supporting a three-line key telephone system.
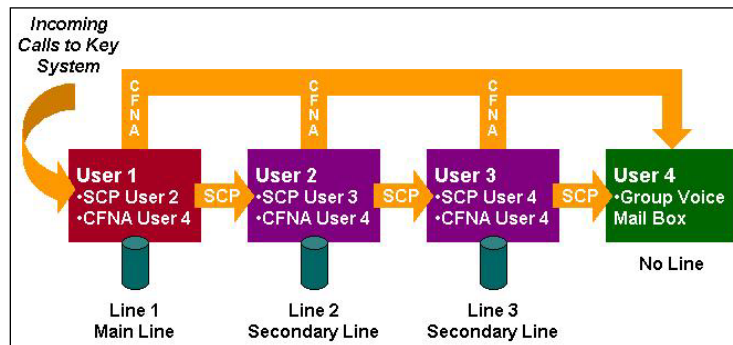
Figure 5 Cisco BroadWorks Configuration for Three-Line KTS

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.

4.27.2 Configuration

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

4.28 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.

4.28.1 Description

The service integrates the virtual private network (VPN) destinations with the Cisco BroadWorks 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.

4.28.2 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.

4.29 VoiceXML

The VoiceXML service allows group administrators to run VXML/CCXML scripts.

4.29.1 Description

The VoiceXML service allows group administrators to create VoiceXML service instances (virtual users) and use them to run VXML/CCXML scripts. VoiceXML virtual users are configured with a device and typical addressing options, such as line/port, public identity (in IMS deployments), directory number, extension, and so on.

The line/port of this device is used to identify the VoiceXML virtual subscriber for VXML script originations. In IMS deployments, the virtual subscriber can also be configured with a public identity, used to identify the virtual subscriber on terminations from the IMS core. The device has a static contact that is used for the media function routing and contains the script name to be executed by the Media Server.

4.29.2 Configuration

The group administrator creates one or more VoiceXML service instances and assigns them a phone number/extension and a device that supports static registrations.

5 Messaging

5.1 Fax Messaging

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

5.1.1 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 email are converted to the TIFF image format and are attached to the email, 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 email account if the user has email notification enabled.

5.1.2 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.

5.2 MWI Delivery to Mobile Endpoint

The MWI Delivery to Mobile Endpoint service notifies the user's mobile device when a new voice message is left for the user. This service is an overlay to the Voice and Fax Messaging services.

5.2.1 Description

With this service, Cisco BroadWorks controls the message waiting indicator on a user's mobile phone on a Global System for Mobile Communications (GSM) network.

When a new message is left for the user, the system sends an MWI message to the specified mobile phone number.

When the user has listened to all their new messages, the system sends a deactivation message.

This service is independent of other messaging notifications and indications.

5.2.2 Configuration

The user can enable or disable the service and specify the mobile phone number where the MWI notification is to be delivered.

5.3 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.

5.3.1 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.

The user can only do post-answer transfers to voice mail.

5.3.2 Configuration

This service has no configuration.

5.4   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.

5.4.1   Description

This service allows Cisco BroadWorks 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 Cisco BroadWorks. Each analog line is mapped to a user account on Cisco BroadWorks and the Voice Mail server is the phone number of a hunt group in Cisco BroadWorks. SMDI Message Desk provides the Voice Mail server with redirection information for incoming calls.
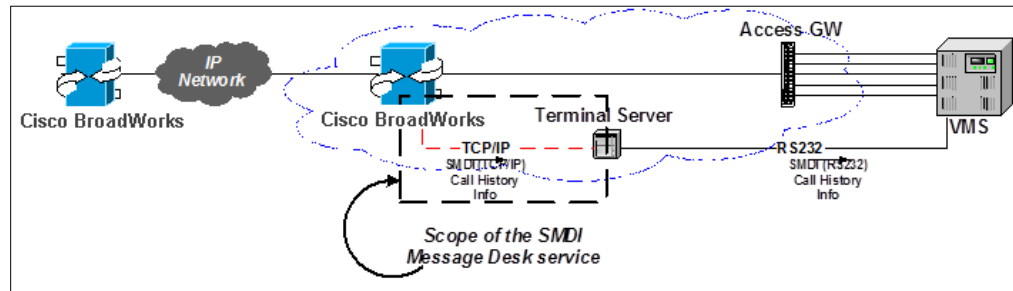
Figure 6 SMDI Message Desk Service

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.

5.4.2   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 it 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.

5.5   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 Cisco BroadWorks with a third-party voice mail platform, while retaining the integration of voice mail with other Cisco BroadWorks services.

5.5.1   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.

5.5.1.1   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.

5.5.1.2   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 system 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.

5.5.1.3   Message Waiting Notification

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

5.5.2   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 the 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. For more information and for the list of service that share this setting, see section *6.14 User No Answer Timer*.

Calls can also be redirected to voice mail using feature access codes. For more information, see sections *3.13 Call Forwarding Always to Voice Mail*, *3.15 Call Forwarding Busy to Voice Mail,* and *3.17 Call Forwarding No Answer to Voice Mail*.

5.6 Voice Messaging

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.

5.6.1 Description

Voice Messaging provides all of the features of a traditional voice messaging solution, plus:

■ Message delivery to any specified email 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 other 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 CommPilot voice portal phone number and have calls sent directly to their mailbox greeting.

The following subsections describe the service in greater detail.

5.6.1.1 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 customized partially or fully by the user.

The caller can then leave a message, press **0** to transfer to an attendant or press ***** to access the voice portal. 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

■ Erase the message and access the voice portal

■ 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.

5.6.1.2 Video Support

Voice 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 email through the voice portal, but only the audio portion is played back.

5.6.1.3 Storage

Messages are stored on standard email servers (POP3, IMAP4, or Microsoft Exchange Server) as *.WAV* or *.MOV* files attached to emails. 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.

5.6.1.4 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 email, can retrieve their voice mail messages from their location, from a third-party location, or from any standard email 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 emails and can be listened to with standard audio software. Messages received can be manipulated similar to any other email (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.

5.6.1.5 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 emailed 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.

5.6.2 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" email address.

The service provider and enterprise administrators can override the system default setting for the *From* field used when sending an email 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.

- Specify the number of rings before the call is forwarded to the user's mailbox. Note, however, that this setting applies to all services with no-answer handling. For more information and for the list of service that share this setting, see section *6.14 User No Answer Timer*.

- 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.

5.7 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.

5.7.1 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 section *5.6 Voice Messaging*):

  – Retrieve messages

  – Compose, forward, or reply to messages

  – Change greetings

- Activate, deactivate, and program their Call Forwarding Always service (see section *3.11 Call Forwarding Always*).

- Select a CommPilot Express profile (see section *3.35 CommPilot Express*).

- 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.)

If auto login is enabled, then 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.

5.7.1.1 Voice Portal Express Mode

The Express Mode provides a simplified interaction between the voice portal and the user to improve the user experience for users who simply want to listen to their messages and do not need advanced features. The express mode provides the following:

- Direct access to the voice mailbox
- Automatic playing and saving of new messages and then moving to the next message
- Disabling of the timestamp message

After accessing the voice portal, the user is placed directly into the voice mailbox. If new messages are present, the first new message is played automatically without user input. The message timestamp prompt is not played for new messages. The IVR commands to save, erase, repeat, and so on are still available and can be entered while the message is being played.

After the new message has played, it is automatically saved and the voice portal moves to the next new message. When all new messages have been played, the voice portal moves to the saved messages and plays the first saved message. After this message is played back to the user, the voice portal prompts the user with the regular save/delete prompt. From this point on, the regular voice portal functionality is used.

If no new or saved messages are present, a voice mail retrieval (VMR) no-message greeting is played, which is the regular functionality.

5.7.1.2 Voice Portal Wizard

The Voice Portal Wizard is (optionally) assigned to groups, and it 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.

5.7.1.3 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 via the CommPilot 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 email 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.

5.7.1.4 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 the user 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.

If the voice portal scope is set to "enterprise", then the 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.

5.7.1.5 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 cannot be directly dialed 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.

5.7.1.6 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 Cisco BroadWorks user ID of the voice portal in such a case. Note that in this case, the group should have a group CLID set.

5.7.1.7 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.

5.7.1.8 The user can make as many calls as desired. Either the user can 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.

5.7.2 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

6 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.

6.1 Alternate User IDs

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

6.1.1 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 Cisco BroadWorks user ID. In such cases, carriers often set up the subscribers' Cisco BroadWorks 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 Cisco BroadWorks.

As Cisco BroadWorks 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 Cisco BroadWorks. 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.

6.1.2 Configuration

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

6.2 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 Cisco BroadWorks services to be offered to CPE users.

6.2.1 Description

The Business Trunking framework enhances the Cisco BroadWorks 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 Cisco BroadWorks.

- The number of users on the CPE may be greater than the number of users on Cisco BroadWorks. As a result, the number of users on Cisco BroadWorks does not provide a reliable indication of the system resources consumed by the CPE. (For example, a single Cisco BroadWorks 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 Cisco BroadWorks, 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 Cisco BroadWorks, 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 Cisco BroadWorks services to be assigned to CPE users as required.

For more information, see the *Cisco BroadWorks SIP Trunking Solution Guide*.

6.2.1.1 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 Cisco BroadWorks 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.

6.2.2 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 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. For more information, see section *6.23 Trunk Bulk Number Provisioning*.

6.3 Call Processing Policies

This feature provides explicit control of certain Cisco BroadWorks call processing behavior.

6.3.1 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.

6.3.1.1 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.

6.3.1.2 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.

6.3.1.3 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. It is strongly recommended to avoid setting that policy to a value that is equivalent to the number of rings before no-answer processing. (For the list of no-answer handling services and setting the number of rings before no-answer processing, see section *6.14 User No Answer Timer*.) 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.

6.3.1.4 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.

6.3.1.5 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.

6.3.1.6 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.

6.3.1.7 Phone List Lookup

This policy allows the administrator to decide whether Cisco BroadWorks 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.

6.3.2 Configuration

Call Processing Policies can be set at the service provider, group, and user levels. For more information, see section *6.3.1 Description*.

6.4 Classmark

This service allows a classmark to be assigned to a user and communicated within SIP messaging between Cisco BroadWorks and the Public Switched Telephone Network (PSTN) during call setup.

6.4.1 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
- BroadWorks 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.

6.4.2 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.

6.5   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.

6.5.1   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.

6.5.1.1   Authorization Code NCOS Capability

This feature adds flexibility to the Communication Barring service, enabling Cisco BroadWorks to apply a Network Class of Service (NCOS) dynamically for originating calls. The system 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.

6.5.2   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.

6.6 CommPilot Web Portals

CommPilot 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.

Cisco BroadWorks provides flexibility in managing service configuration using web portals. The web portals (called CommPilot 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 section *6.16 Restricted Administrative Access*.

Cisco BroadWorks provides a different CommPilot web portal or interface for each role:

- The system provider administrator uses the CommPilot web portal to establish system-wide settings, such as setting messages callers hear when they interact with Cisco BroadWorks 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 CommPilot web portal to perform a limited number of functions related to their department.

- End users use the user version of the CommPilot web portal to configure their services.

6.7 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.

6.7.1 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 necessarily 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.

6.7.2 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
52	Anonymous Call Rejection Interrogation
#8	Automatic Callback Deactivation
#9	Automatic Callback Menu Access
*14	BroadWorks 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

Code	Feature
#76	Call Forwarding Selective Activation
#77	Call Forwarding Selective Deactivation
54	Calling Line ID Delivery Blocking Interrogation
*67	Calling Line ID Delivery Blocking per Call
*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
*79	Do Not Disturb Deactivation
#83	Escalate Call to Supervisor
#63	Executive-Assistant Call Push

Code	Feature
#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
#28	Mobility Calling Line ID Activation
*28	Mobility Calling Line ID Activation Per Call
#29	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

Code	Feature
*85	Number Portability Announcement Deactivation
*71	Per-Call Account Code
#0322	Push Notification Retrieval
*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

6.8 Direct Route

The Direct Route service enables the system to route network-originated calls directly to a Private Branch Exchange (PBX) identified by a trunk group identifier without identifying a Cisco BroadWorks user from a lookup of the called number.

6.8.1 Description

The Direct Route service enables Cisco BroadWorks to identify a route to a destination PBX. To identify the route, the Application Server parses the trunk group identifier in the Request-URI of an incoming INVITE request and tries to match it to a direct route identifier. If it finds a match, then it selects the user to whom the direct route identifier is assigned and routes the call using that user's routing profile. The user must be a trunking user, and the Application Server routes the call as a trunk group call.

When a user is assigned the Direct Route service, then that user should be considered a "special purpose" user. While the Application Server does not enforce any special provisioning restrictions on a direct route user, there are some differences in the way the Application Server performs the call processing.

The Application Server applies a one-call-per-session policy for a direct route user's calls, both terminating and originating. That is, the Application Server processes the calls in unmapped sessions. This allows processing of direct route calls with greater concurrency, which improves call throughput for high call volume. Thus, this policy is appropriate for, example, an Interactive Voice Response (IVR) system in the enterprise. However, because the sessions are unmapped, the Application Server cannot provide certain services that involve two or more calls, such as Three-Way Calling.

For more information, see the *Cisco BroadWorks Service Interaction Guide*.

6.8.1.1 Freephone Call Termination Example

The Direct Route service allows Cisco BroadWorks to route an incoming call from the Public Switched Telephone Network (PSTN) to a PBX based on a route identifier selected by the network device. This service can be used by a network operator to support a freephone service.

The following figure shows how a network operator can implement a freephone service. This scenario begins when a class four softswitch receives an incoming call to a freephone (toll-free) number. Following its usual procedure, the switch queries the freephone number database for translations. The database returns a number and a trunk group identifier. The switch creates an INVITE request with the translated number and parameters with values that identify the destination PBX. The switch sends this INVITE request to a Cisco BroadWorks Network Server, which performs a lookup of the trunk group identifier, selects the appropriate Cisco BroadWorks Application Server for the call, and sends a 302 response to the switch. Upon receiving the response, the switch sends the INVITE request to the Application Server, which selects a trunk group for routing and sends an outgoing INVITE request to that trunk group. The Application Server does not translate the destination number in the received request; however, it does relay that number to the outgoing INVITE request. The PBX receives the INVITE request and handles it appropriately.
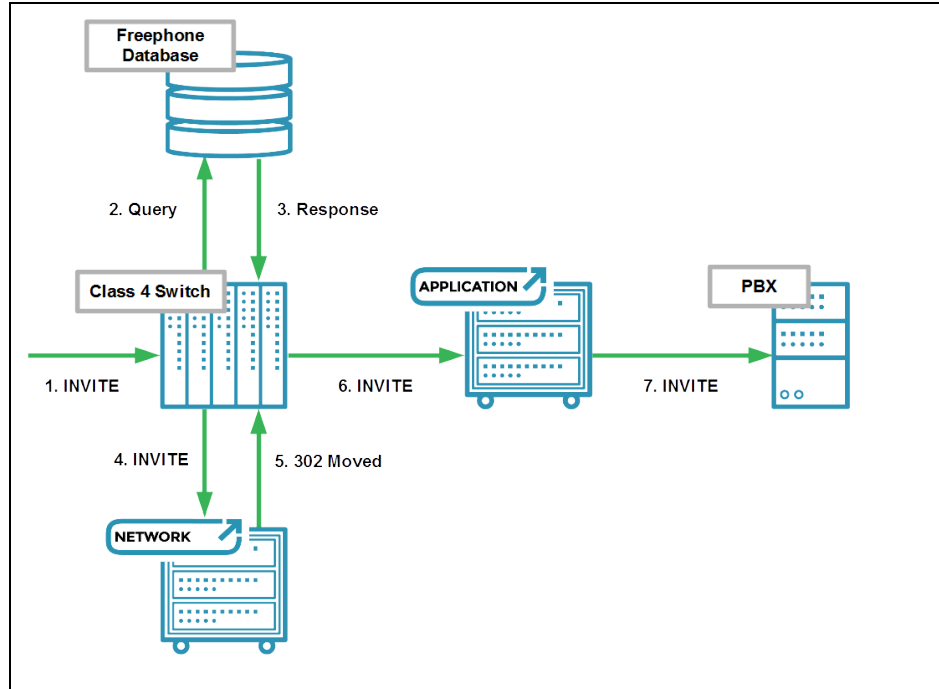
Figure 7 Termination of Freephone Call

The Direct Route service is essential for this implementation to work. The freephone number database returns a translated number that is not provisioned in the Application Server. Nevertheless, the Application Server is able to route the call based on the provided parameter values.

6.8.2 Configuration

The administrator assigns direct route identifiers to the user's Direct Route service.

There are two types of direct route identifiers. They are distinguished by how the direct route identifier is used within the SIP signaling.

■   The trunk identity type has a label component and a domain component, and these are used as the values of the *tgrp* and *trunk-context* parameters, respectively.

■   The DTG type has only a single component, and this is used as the value of the *dtg* parameter.

A trunk identity type direct route identifier is similar to a trunk group identity that is provisioned for a trunk group, in that both of these identities may appear as values of the *tgrp* and *trunk-context* parameters. Similarly, the DTG type direct route identifier is similar to the OTG/DTG Identity in that both may appear as the value of the *dtg* parameter. However, within Cisco BroadWorks, these are unrelated resources. For example, it is possible for the identifier "trunk123" to be provisioned both as a trunk group's OTG/DTG Identity and as a DTG type direct route identifier.

The Direct Route service is effective only if it is assigned to a trunking user.

A direct route user, that is, a trunking user to whom the Direct Route service is assigned, is not required to have a phone number assigned. However, if the direct route user does not have a phone number, then that user must have an extension and the user's group must have a group calling line ID (CLID).

6.9 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.

6.9.1 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. *Figure 8* shows a diagram of a possible Enterprise Trunking deployment.
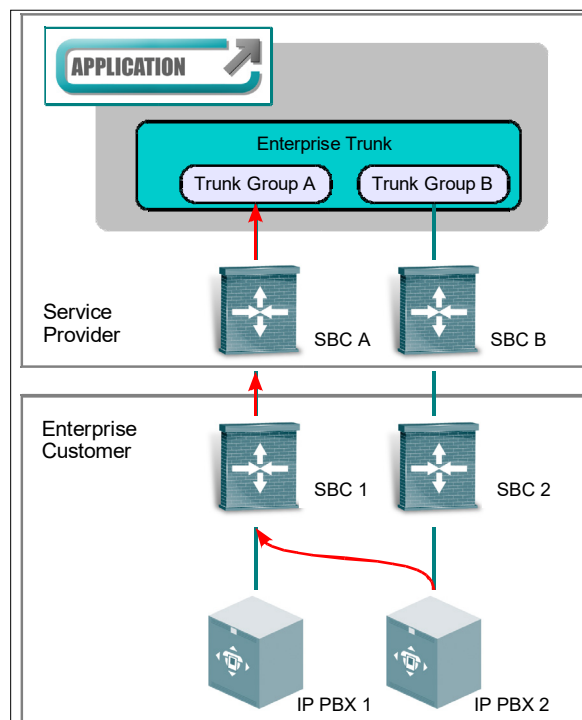
Figure 8 Example of Enterprise Trunking Deployment

6.9.2 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.

6.10 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.

6.10.1 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.

6.10.2 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.

6.11 Immediate SIP Connectivity for New Site (Route List)

Cisco BroadWorks allows an operator to provide immediate SIP trunking connectivity, including direct inward dialing, to all users on a Private Branch Exchange (PBX) via a single Cisco BroadWorks user profile.

6.11.1 Description

Cisco BroadWorks allows an operator to bring a new site online simply by provisioning a range of directory numbers (DNs) or number prefixes for the site. This capability provides connectivity for both outbound and inbound SIP trunking calls. To support the capability, Cisco BroadWorks offers a user-licensable service called Route List and resources called enterprise trunk number range and an enterprise trunk number prefix. Route List is used to associate a DN range (enterprise trunk number range) or prefix (enterprise trunk number prefix) with an end-customer site. Enterprise trunk number ranges and enterprise trunk number prefixes play similar role, but the prefixes allow end users to have variable-length phone numbers.

After provisioning is completed, the new site is able to make outbound calls and receive inbound calls using the numbers contained in the enterprise trunk number range or matching one of the assigned enterprise trunk number prefixes. The system processes such calls using the service profile of the Route List user. Because the service profile is shared among multiple users on the PBX, the Route List user should not have user-centric services, such as Voice Messaging. However, the shared user profile does provide a way to apply connectivity-related functionality, such as Call Screening. Furthermore, the system may apply general SIP trunking functionality, such as calling line identity selection, call capacity management, and so on.

When a customer wants to upgrade some users to advanced services, the operator creates an individual Cisco BroadWorks user profile for each user that requires advanced services. To migrate a user to an individual profile, the operator creates a new enterprise trunk user and assigns that user a DN. The individual DN assignment has precedence over the Route List enterprise trunk number range and enterprise trunk number prefix assignment. Therefore, the operator does not need to make changes to the Route List configuration and non-migrated users continue to be processed using the service profile of the Route List user.

To downgrade a user back from advanced to basic services, the operator deletes the user's individual service profile. After the user's profile is deleted, if the user's DN is covered by the enterprise trunk number range or enterprise trunk number prefix assigned to the Route List user, the system processes the user's calls with the Route List user's service profile.

6.11.2 Configuration

To bring a new site online via SIP trunking, an administrator needs to perform the following provisioning steps on Cisco BroadWorks:

1) Get the Route List service license and authorize the service to the group.

2) Create a new device profile for the PBX, create one or more trunk groups, and create an enterprise trunk.

3) Create a new trunking user and assign the user to the enterprise trunk. Alternatively, select one of the trunk group pilot users (who must also be assigned to the enterprise trunk).

4) Provision an enterprise trunk number range or enterprise trunk number prefix for the numbers that are to be assigned to the site.

5) Assign the Route List service to the trunking user.

6) Assign the enterprise trunk number range or enterprise trunk number prefix to the Route List service of the trunking user.

6.12 Large Enterprise Support

Cisco BroadWorks provides support for large multisite enterprises, with the enterprise layer in the Cisco BroadWorks provisioning model. This layer allows the Cisco 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.

6.12.1 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 could 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.

6.12.2 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.
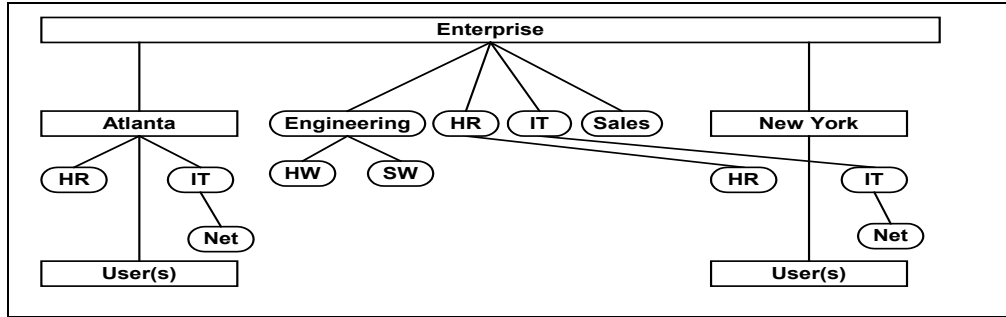
Figure 9 Departments

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.

6.12.3 Enterprise-wide Group Services

The Cisco BroadWorks 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 information, see section *5.7.1.4 Voice Portal Scope*.

6.12.4 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.

6.13 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.

6.13.1 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.

6.13.2 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 or virtual user, that is, a user can be 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.

6.14 User No Answer Timer

The user no-answer timer is used by the user no-answer handling services and is shared by all these services.

6.14.1 Description

The user no-answer timer is shared by the user services with no-answer handling. The service is not invoked until the no-answer timer expires. Note that a service cannot be invoked even if the timer expires, if another service or feature processes the call first.

The user can set the timer from any no-answer handling service, but the timer is shared by all no-answer handling services. Therefore, changing the timer in one service changes it for all these services. Note that the modification applies even if the service where the timer is changed is not active.

Following are the services that share the no-answer timer, in order of precedence:

■ Personal Assistant

■ Sequential Ring (the timer applies only to the base location)

■ Call Forwarding No Answer

■ Voice Messaging User

■ Third-Party Voice Mail Support

6.14.2 Configuration

The system administrator can set the minimum value to which the no-answer timer can be set and the default value of the timer. "None" is also allowed independently of the minimum configured.

The user can configure their no-answer timer either through the web portal or by using the feature access code as follows:

■ From the *User – Personal Assistant* web page by configuring the *Number of rings* setting.

■ From the *User – Sequential Ring* web page by configuring the *Number of rings for Base Location* setting.

■ From the User – *Call Forwarding Now Answer* web page by configuring the *Number of rings before forwarding* setting.

■ From the *User – Greetings* web page by setting the *Number of rings before greeting* setting. This page allows the user to configure various greetings for their Voice Messaging service.

■ From the *User – Third-Party Voice Mail Support* web page by setting the *Number of rings before greeting* setting.

■ By dialing the No Answer Timer feature access code (default *610).

It is strongly recommended to avoid setting the number of rings 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.

6.15 Resource Priority

This service allows an administrator to assign Q735 resource priority to the user.

6.15.1 Description

The SIP *Resource-Priority* header is used for the Multi-Level Precedence and Preemption (MLPP) functionality to assign a priority to a call. This priority is used when network congestion is present to rank incoming calls based on priority and to preempt calls with a lower priority so that the higher priority calls can proceed.

Resource Priority is an origination service. This service allows an administrator to assign Q735 resource priority to the user. Additionally, a default resource priority can be provisioned at the system level.

When a Cisco BroadWorks user has a resource priority assigned and a call originating from the user terminates to a network destination, the user's resource priority is sent to the network destination. If the Resource Priority service is not assigned to the user or if the service is configured to use the default resource priority, the default resource priority is sent to the network destination.

6.15.2 Configuration

Group administrators can configure the value for resource priority for the user, which can take one of the following values: q735.0, q735.1, q735.2, q735.3, or q735.4.

6.16 Restricted Administrative Access

This feature provides granularity to the authorization of each administrative level of the CommPilot web portal. Cisco BroadWorks 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.

6.16.1 System-level Policies – Password Rules

The system administrator defines rules of creating and updating passwords at the system level and has the option to:

- 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 Cisco BroadWorks. 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.

6.16.2 Provisioning Administration-level Policies

6.16.2.1 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.

6.16.2.2 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.

6.16.2.3 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.

6.16.3 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.

6.16.4 Configuration

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

6.17 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.

6.17.1 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, BroadWorks 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.

> **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.

6.17.2 Configuration

Each user or administrator defines schedules at their administrative level.

The system administrator configures the system-wide *emptyHolidayScheduleIsOutOfSchedule* setting. If set to "true", then an empty holiday schedule never applies. If set to "false", then an empty holiday schedule always applies.

6.18 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.

6.18.1 Description

Selective criteria are used to screen calls in selective services, such as Call Forwarding Selective or BroadWorks 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

6.18.2 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, see the section for that service.

6.18.3 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.

6.19 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).

6.19.1 Description

There are many instances in which a carrier would want 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 Cisco BroadWorks, 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.

6.19.1.1 Session Admission Control Based on Codec Weight

Session Admission Control allows assigning weight to audio codecs to count sessions that originate from or terminate to a site with limited bandwidth. Since some codecs can use a lot more bandwidth than others, there may be a need to control the contribution of each session to the session count, based on the codec proposed and used.

An enhancement to Session Admission Control introduces a configurable codec weight per codec configurable for each SAC group. The system administrator can define groups of media codecs with their associated weights, which can then be linked to SAC groups.

Depending on the system configuration, the preferred codec or the codec with highest weight (and their associated weight) is used for each call.

6.19.1.2 Subscriber-level Session Admission Control Granularity

Session Admission Control allows to configure whether a call to or from a network location type is counted against a subscriber's session count limits.

A customer may want to process BroadWorks Mobility locations within the enterprise/service provider while they process BroadWorks Anywhere and Remote Office outside the enterprise/service provider for subscriber-level Session Admission Control functionality. For these reasons, an enhancement to the Session Admission Control provides the following functionality to the subscriber-based Session Admission Control functionality at the enterprise/service provider and group levels.

■ Configure whether an additional originating subscriber session is consumed when a BroadWorks Anywhere location is involved in a call. It includes calls connected via the BroadWorks Anywhere Portal.

- Configure whether an additional originating subscriber session is consumed when a Remote Office network location is involved in a call.

- Does not consume enterprise-level session resources when a Cisco BroadWorks user initiates a call to a BroadWorks Mobility number if the call is processed as a network call and both calling and called users belong to the same enterprise/service provider.

- Does not consume group-level session resources when a Cisco BroadWorks user initiates a call to a BroadWorks Mobility number if the call is processed as a network call and both calling and called users belong to the same group.

- Rejects the call if a network location is configured to consume a subscriber session and there are no more subscriber sessions available to alert the network location. This provides a consistent user experience to the calling and called users, preventing inconsistent behavior where a user's network location may sometimes ring and at other times not ring due to Session Admission Control capacity.

6.19.2 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.

6.19.2.1 Session Admission Control Based on Codec Weight

The system administrator configures the following:

- Codec and associated codec weight to use in the Session Description Protocol offer prior to call answer (preferred codec or codec with highest weight).

- Media groups, codecs and their associated weights. Media groups can then be assigned the service provider/enterprise or group SAC groups.

The service provider/enterprise or group administrator associate media groups with SAC groups.

6.19.2.2 Subscriber-level Session Admission Control Granularity

The system administrator can enable or disable the subscriber-level Session Admission Control granularity functions. This configuration only applies when the enterprise/service provider is configured not to count sessions within their organization in the total number of sessions.

- Decide whether to include an origination/termination from a Cisco BroadWorks user to another Cisco BroadWorks user's mobile number within the service provider/enterprise against the user's originating/terminating session count.

■ Decide whether an additional originating resource is consumed when a BroadWorks Anywhere location is involved in a call.

■ Decide whether an additional originating resource is consumed when the Remote Office service alerts the user's remote location.

6.20 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.

6.20.1 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 affect 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.

6.20.1.1 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 Cisco BroadWorks; 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.

6.20.1.2 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.

6.20.2 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.

6.21 Terminating Alternate Trunk Identity

This service allows an administrator to provision a trunking user with a terminating address that has a special purpose. The address has no restrictions for uniqueness. As a result, the same address is allowed to be assigned to more than one user in the system.

6.21.1 Description

Terminating Alternate Trunk Identity is a user service assignable to trunking users. It allows an administrator to provision a trunking user with a non-unique trunk identity used for terminating calls to the user. When the Application Server terminates a call to a trunking user with a provisioned terminating identity, the identity is used to form the *Request-URI* and *To* headers in the SIP request based on the trunking mode in use.

This value is only used when terminating to the primary endpoint and is not used when terminating to endpoints defined by services such as Shared Call Appearance. This value is used exactly as it is provisioned. For example, if the device endpoint is not E.164 capable and the terminating trunk identity is provisioned in E.164 format, the E.164 format is sent to the device anyway.

The terminating trunk identity is similar to the alternate trunk identity, but is used for terminating calls only and has a higher precedence than the alternate trunk identity. When both are provisioned, the terminating trunk identity is used in places where the alternate trunk identity is usually used.

6.21.1.1   Example of Using Terminating Trunk Identity

A customer network has different Private Branch Exchanges (PBXs) that can handle the same set of numbers. The customer enables the trunk group prefix and provisions a unique value for it. When a call terminates to this trunk group, this unique trunk group prefix is combined with the provisioned non-unique trunk identity to form a unique routable address to a session border controller (SBC). The SBC strips out the unique trunk group prefix and uses it to identify the provisioned PBX. The SBC then routes the non-unique trunk identity to the destination PBX.
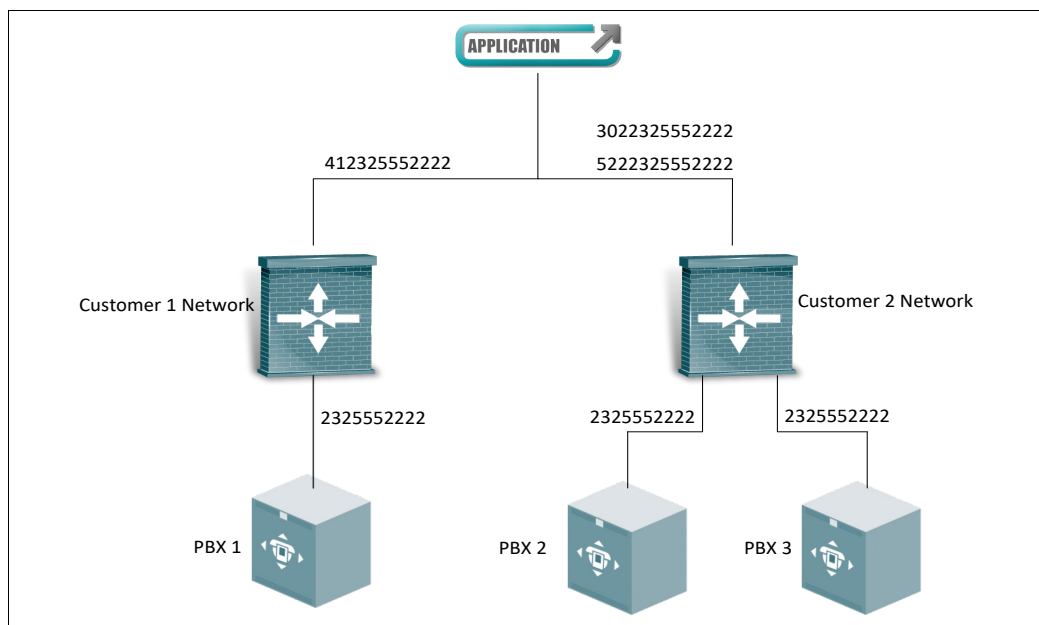
Figure 10 Example of Using Terminating Trunk Identity

6.21.2 Configuration

The group administrator provisions a terminating trunk identity to the user. The identity can have the same value as an existing alternate trunk identity or another terminating trunk identity.

6.22 Shared Devices

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

6.22.1 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 Cisco BroadWorks is used to provide voice mail for another host system (in which case the "host" system is configured as a shared access device for delivering the message waiting indicator notifications).

6.22.2 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.

6.23 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.

6.23.1 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.

6.23.2 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.

6.24 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.

6.24.1 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 Cisco BroadWorks 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 CommPilot 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.

6.24.2 Configuration

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

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