PacketCable™

Business SIP Services (BSS) Feature Description Technical Report

PKT-TR-BSS-C01-140314

CLOSED

Notice

DISCLAIMER

Document Status Sheet

Document Control Number:	PKT-TR-BSS-C01-140314
Document Title:	Business SIP Services (BSS) Feature Description Technical Report
Revision History:	V01 - Issued September 5, 2008
	C01 - Closed March 14, 2014
Date:	March 14, 2014

Abstract

This technical report describes the PacketCable Business SIP Services features and underlying assumptions. Business SIP Services includes many of the features commonly available to residential subscribers of public switched telephone network services. Examples include Caller ID, Call Waiting, Speed Dialing, Call Forwarding, Call Blocking, Call Transfer, Do Not Disturb, and Emergency calling capabilities. A number of additional telephony features specific to business environments are also included in this offering. Examples of these features include Shared Call Appearance, Group Services such as Hunt Groups, Private Numbering Plan, Call Pickup, Simultaneous Ring, and Sequential Ring. In addition to the voice-oriented features, a number of desktop-oriented features such as Web Portal and Desktop Integration are considered useful for businesses; thus they are within the scope of Business SIP Services. This technical report also describes the expectations for the devices that can be used for Business SIP Services.

Table of Contents

Figures

.

1 INTRODUCTION

1.1 PacketCable Business SIP Services Overview

The Business SIP Services (BSS) have been identified by CableLabs and its member companies as an important set of voice services for Small and Medium sized Businesses (SMBs). Small business is generally defined to be a business with less than 10 lines of service. Medium businesses generally have from 10 to 99 lines of service.

This document outlines the scope and description of features to be supported by Business SIP Services. The detailed description of features in this report forms the basis for the BSS specification development, where the technical requirements are provided to define the behavior of Session Initiation Protocol (SIP) clients that interact with the application servers and call control elements in a PacketCable network to implement the features.

NOTE: All references to PacketCable in this document imply PacketCable 2.0 unless specified otherwise.

1.2 PacketCable Business SIP Services Motivation

The cable industry has achieved great success in deploying residential voice services based on the PacketCable VoIP architecture. Following this success, the industry is also developing PacketCable 2.0, a next-generation platform for multimedia services over cable. This new platform can be used for residential voice services (e.g., Residential SIP Telephony), and is also capable of supporting the voice communication needs of millions of small and medium-sized businesses in North America. Over the next few years, this segment of businesses represents a significant revenue opportunity for the cable industry.

Addressing the diverse needs of the SMB market may require cable operators to have multiple voice products, differentiated by access network methodology (fiber or coax), client types (embedded in a cable modem or a stand-alone IP device), and type of services offered (cable hosted or customer hosted). The intent of this technical report is to describe common SMB features and services that could be offered in a cable hosted solution. The features are described from the customer perspective, independent of the physical service delivery architecture and network topology. The feature descriptions do not imply any specific implementation approach.

2 REFERENCES

This technical report uses the following references.

2.1 Informative References

[ARCH-FRM TR] PacketCable Architecture Framework Technical Report, PKT-TR-ARCH-FRM-C01-140314, March 14, 2014, Cable Television Laboratories Inc.

[CMSS1.5] PacketCable CMS to CMS Signaling 1.5 Specification, PKT-SP-CMSS1.5-I07-120412-April 12, 2012, Cable Television Laboratories Inc.

[CODEC-MEDIA] PacketCable Codec-Media Specification, PKT-SP-CODEC-MEDIA-C01-140314, March 14, 2014, Cable Television Laboratories Inc.

[CSTA] ECMA-269, Services for Computer Supported Telecommunications Applications (CSTA) Phase III, 7th edition (December 2006), ECMA.

[DSML v2] OASIS Service Provisioning Markup Language (SPML) v2 - DSML v2 Profile, April 2006, OASIS Standard.

[EG 201 188] ETSI EG 201 188, Public Switched Telephone Network (PSTN Network Termination Point (NTP) analogue interface, January 2000.

[EN 300 001] ETSI EN 300 001, Attachments to the Public Switched Telephone Network (PSTN), October 1995.

[EN 300 659-1] ETSI EN 300 659-1, Access and Terminals (AT); Analogue access to the Public Switched Telephone Network (PSTN); Subscriber line protocol over the local loop for display (and related) services; Part 1: On-hook data transmission, January 2001.

[EN 300 659-3] ETSI EN 300 659-3, Access and Terminals (AT); Analogue access to the Public Switched Telephone Network (PSTN); Subscriber line protocol over the local loop for display (and related) services; Part 3: Data link message and parameter codings, January 2001.

[GR 30] Telcordia GR-30, LSSGR: Voiceband Data Transmission Interface (FSD 05-01-0100), December 1998, FR-64.

[GR 31] Telcordia GR-31, LSSGR: CLASS Feature: Calling Number Delivery (FSD 01-02-1051), June 2000, FR-64.

[GR 215] Telcordia GR-215, LSSGR: CLASS Feature: Automatic Callback (FSD 01-02-1250), June 2000, FR-64.

[GR 216] Telcordia GR-216, CLASS Feature: Customer Originated Call Trace (FSD 01-02-1052), June 2000, FR-64.

[GR 217] Telcordia GR-217, LSSGR: CLASS Feature: Selective Call Forwarding (FSD 01-02-1410), June 2000, FR-64.

[GR 218] Telcordia GR-218, LSSGR: CLASS Feature: Selective Call Rejection (FSD 01-02-0760), May 1992, FR-64.

[GR 220] Telcordia GR-220, LSSGR: CLASS Feature: Screening List Editing (FSD 30-28-0000), June 2000, FR-64.

[GR 227] Telcordia GR-227, LSSGR: CLASS Feature: Automatic Recall (FSD 01-02-1260), June 2000, FR-64.

[GR 391] Telcordia GR-391, LSSGR: CLASS Feature: Calling Identity Delivery Blocking Features (FSD 01-02-1053), June 2000, FR-64.

[GR 505] Telcordia GR-505, LSSGR: Call Processing, December 2006, FR-64.

[GR 506] Telcordia GR-506, LSSGR: Signaling for Analog Interfaces, November 1996, FR-64.

[GR 524] Telcordia GR-524, LSSGR: Attendant and Customer Switching System Features and Customer Interfaces (FSDs 04-01-0000, 04-02-0000, 04-05-0000), July 1987, FR-64.

[GR 531] Telcordia GR-531, LSSGR: Interoffice (FSDs 25-05-0903, 25-06-0501, 25-06-0502, 25-06-0506), June 2000, FR-64.

[GR 562] Telcordia GR-562, LSSGR: Manual Line Features (FSD 01-02-0301), June 2000, FR-64.

[GR 564] Telcordia GR-564, LSSGR: Code Restriction and Diversion (FSD 01-02-0600), June 2000, FR-64.

[GR 567] Telcordia GR-567, LSSGR: CLASS Feature: Anonymous Call Rejection (FSD 01-02-1060), June 2000, FR-64.

[GR 568] Telcordia GR-568, LSSGR: Series Completion (FSD 01-02-0801), May 1990, FR-64.

[GR 569] Telcordia GR-569, LSSGR: Multiline Hunt Service (FSD 01-02-0802), May 1990, FR-64.

[GR 570] Telcordia GR-570, LSSGR: Speed Calling (FSD 01-02-1101), June 2000, FR-64.

[GR 571] Telcordia GR-571, LSSGR: Call Waiting (FSD 01-02-1201), June 2000, FR-64.

[GR 572] Telcordia GR-572, LSSGR: Cancel Call Waiting (FSD 01-02-1204), June 2000, FR-64.

[GR 577] Telcordia GR-577, LSSGR: Three-Way Calling (FSD 01-02-1301), June 2000, FR-64.

[GR 578] Telcordia GR-578, LSSGR: Sensitive Three-Way Calling (FSD 01-02-1304), June 2000, FR-64.

[GR 579] Telcordia GR-579, LSSGR: Add-On Transfer and Conference Calling Feature (FSD 01-02-1305), June 2000, FR-64.

[GR 580] Telcordia GR-580, LSSGR: Call Forwarding Variable (FSD 01-02-1401), June 2000, FR-64.

[GR 586] Telcordia GR-586, LSSGR: Call Forwarding Subfeatures (FSD 01-02-1450), June 2000, FR-64.

[GR 590] Telcordia GR-590, LSSGR: Call Pickup Features (FSD 01-02-2800), July 1989, FR-64.

[GR 605] Telcordia GR-605, LSSGR: Authorization Codes for Automatic Flexible Routing (AFR) and Account Codes for Basic Business Group and AFR (FSD 02-02-1010), April 1991, FR-64.

[GR 1176] Telcordia GR-1176, OSSGR: Custom Call-Handling Features (FSD 80 Series), A Module of OSSGR, March, 1999, FR-271.

[GR 1188] Telcordia GR-1188, LSSGR: CLASS Feature: Calling Name Delivery Generic Requirements (FSD 01-02-1070), December 2000, FR-64.

[GR 1401] Telcordia GR 1401, LSSGR: Visual Message Waiting Indicator Generic Requirements (FSD 01-02-2000), June 2000, FR-64.

[GR 2913] Telcordia GR 2913-CORE, LSSGR: Generic Requirements for Call Park (FSD 01-02-2400), February 1996, FR-64.

[GR 2917] Telcordia GR 2917, LSSGR: Generic Requirements for ISDN Smart Attendant Service, September 1996.

[ITU-T E.164] ITU-T Recommendation E.164, The international public telecommunication numbering plan, February 2005.

[ITU-T T.30] ITU-T Recommendation T.30, Procedures for document facsimile transmission in the general switched telephone network, September 2005.

[ITU-T T.38] ITU-T Recommendation T.38, Procedures for real-time Group 3 facsimile communication over IP networks, September 2005.

[ITU-T V.8] ITU-T Recommendation V.8, Procedures for starting sessions of data transmission over the public switched telephone network, November 2000.

[ITU-T V.18] ITU-T Recommendation V.18, Operational and interworking requirements for DCEs operating in the text telephone mode, November 2000.

[ITU-T V.21] ITU-T Recommendation V.21, 300 bits per second duplex modem standardized for use in the general switched telephone network, November 1988.

[ITU-T V.25] ITU-T Recommendation V.25, Automatic answering equipment and general procedures for automatic calling equipment on the general switched telephone network including procedures for disabling of echo control devices for both manually and automatically established calls, October 1996.

[ITU-T V.152] ITU-T Recommendation V.152, Procedures for supporting voice-band data over IP networks, January 2005.

[OB 781] Operational Bulletin No. 781 (1.II.2003) and Annexed List: Various tones used in national networks (According to ITU-T Recommendation E.180, 03/1998) (Position on 1 February 2003).

[Parlay X] Parlay X Web Service Specifications, version 3.0. Published by ETSI as ES 202 504 V1.1.1 and 3GPP as TS 29.199.

[PKT 24.229] PacketCable Release Sip and SDP Stage 3 Specification 3GPP TS 24.229, PKT-SP-24.229-C01-140314, March 14, 2014, Cable Television Laboratories, Inc.

[PKT 33.203]	PacketCable Release 3G Security; Access Security for IP-based services Specification 3GPP TS 33.203, PKT-SP-33.203-C01-140314, March 14, 2014, Cable Television Laboratories, Inc.
[RFC 3261]	IETF RFC 3261, SIP: Session Initiation Protocol, June 2002.
[RFC 3725]	IETF RFC 3725, Best Current Practices for Third Party Call Control (3pcc) in the Session Initiated Protocol (SIP), April 2004.
[RFC 3892]	IETF RFC 3892, The Session Initiation Protocol (SIP) Referred-By Mechanism, September 2004.
[RFC 3959]	IETF RFC 3959, The Early Session Disposition Type for the Session Initiation Protocol (SIP), December 2004.
[RFC 4510]	IETF RFC 4510, Lightweight Directory Access Protocol (LDAP): Technical Specification Road Map, June 2006.
[RFC 4733]	IETF RFC 4733, RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals, December 2006.
[RST E-DVA]	PacketCable Residential SIP Telephony E-DVA Specification, PKT-SP-RST-E-DVA-C01-140314, March 14, 2014, Cable Television Laboratories, Inc.
[RSTF]	PacketCable Residential SIP Telephony Feature Specification, PKT-SP-RSTF-C01-140314, March 14, 2014, Cable Television Laboratories, Inc.
[RST TR]	Residential SIP Telephony Feature Definition Technical Report, PKT-TR-RST-C01-140314, March 14, 2014, Cable Television Laboratories, Inc.
[SAML]	Security Association Markup Language (SAML) V2.0 specification suite, Approved on 15 March 2005, OASIS.
[SOAP]	W3C Recommendation Simple Object Access Protocol (SOAP) Version 1.2, Second Edition, 27 April 2007, W3C.
[SR-TSV-002476]	CPE Compatibility Considerations for the Voiceband Data Transmission Interface, December 1992, Telcordia.
[TR 23.892]	3GPP TR 23.892, IP Multimedia System (IMS) centralized services, Release 8, V8.0.1 (2008-03).
[TS 23.292]	3GPP TS 23.292, IP Multimedia System (IMS) Centralized Services; Stage 2, Release 8, V8.0.0 (2008-06).

2.2 Reference Acquisition

- Cable Television Laboratories, Inc., 858 Coal Creek Circle, Louisville, CO 80027; Phone +1-303-661-9100; Fax +1-303-661-9199; http://www.cablelabs.com
- Internet Engineering Task Force (IETF) Secretariat, 46000 Center Oak Plaza, Sterling, VA 20166, Phone +1-571-434-3500, Fax +1-571-434-3535,: http://www.ietf.org/
- Telcordia Technologies Customer Service, Piscataway, New Jersey 08854-4156, USA http://telecom-info.telcordia.com/site-cgi/ido/index.html
- ETSI, European Telecommunications Standards Institute, F-06921 Sophia Antipolis Cedex, France, Tel. +33-4-92-94-42-00, Fax +33 4 93 65 47 16, http://www.etsi.org/
- ITU-T, International Telecommunication Union (ITU), Place des Nations, 1211 Geneva 20, Switzerland, +41 22 730 5111, http://www.itu.int/home/
- Organization for the Advancement of Structured Information Standards (OASIS), Post Office Box 455, Billerica, MA 01821, USA, +1 978 667 5115, http://www.oasis-open.org
- ECMA International, Rue du Rhone 114, CH-1204, Geneva, Switzerland, +41 22 849 6001, http://www.ecma-international.org

3 TERMS AND DEFINITIONS

This Technical Report uses the following terms and definitions:

Business Group	The group of users belonging to the small/medium business. All these users must be served by the PacketCable Business SIP Services network. Users belonging to a Business Group may be geographically distributed. A user can only be a member of one Business Group.
Business Group Administrator	A person who is empowered to administer calling features and services for the Business Group, for example through a Web Portal. The Administrator is typically a nominated employee of the small/medium business.
Media Gateway	Devices bridging between the PacketCable IP Voice Communication network and the PSTN. A Media Gateway provides the bearer circuit interfaces to the PSTN and transcodes the media stream.
Off Hook	The active state of a traditional telephone, while a call is in progress or being attempted, and the telephone handset is out of its cradle.
On Hook	The idle state of a traditional telephone, while no call is in progress and the telephone handset is sitting in its cradle.
Public User Identity	The globally unique public address of a user (defined as Address-of-Record in RFC 3261).
SIP Client	The functional element used by subscribers to attach to the PacketCable network.

4 ABBREVIATIONS AND ACRONYMS

This Technical Reports use the following abbreviations and acronyms:

1PCC	First Party Call Control
3PCC	Third Party Call Control
3WC	Three Way Calling
AC	Auto Callback or Alternating Current
ACR	Anonymous Call Rejection
AR	Auto Recall
AS	Application Server
BCT	Blind Call Transfer
BG	Business Group
BLV	Busy Line Verification
BSS	Business SIP Services
CAMEL	Customized Applications for Mobile networks Enhanced Logic
CCT	Consultative Call Transfer
CCW	Cancel Call Waiting
CFBL	Call Forwarding Busy Line
CFDA	Call Forwarding Don't Answer
CFNA	Call Forwarding Not Available
CFV	Call Forwarding Variable
CIDS	Caller Identity Delivery and Suppression
CIDS-D	Caller Identity Delivery and Suppression-Delivery
CIDS-S	Caller Identity Delivery and Suppression-Suppression
CMTS	Cable Modem Termination System
CNAM	Calling Name
CND	Calling Number Delivery
COS	Class Of Service
CPE	Customer Premise Equipment
CPU	Group Call Pickup
CT	Call Transfer
CTD	Click To Dial
CW	Call Waiting
DID	Direct Inward Dialing
DN	Directory Number

DND	Do Not Disturb
DNH	Directory Number Hunting
DOD	Direct Outward Dialing
DPN	Directed Call Pickup
DPU	Directed Call Pickup with Barge-In
DSCP	Differentiated Services Code Point
DTMF	Dual Tone Multi Frequency
DVA	Digital Voice Adapter
ECT	Explicit Call Transfer
E-DVA	Embedded Digital Voice Adapter
FGD	Feature Group D
HSS	Home Subscriber Server
IC/INC	Inter LATA Carrier/International Carrier
ICS	IMS Centralized Services
IDDD	International Distance Direct Dialing
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ITU	International Telecommunication Union
IVR	Interactive Voice Response system
LAN	Local Area Network
LATA	Local Access and Transport Area
LDAP	Lightweight Directory Access Protocol
LIDB	Line Identification Database
MGC	Media Gateway Controller
MLHG	Multi-Lined Hunt Group
MWI	Message Waiting Indicator
NAT	Network Address Translation
OCB	Outbound Call Blocking
OCR	Outbound Call Restriction
PDA	Personal Digital Assistant
PIN	Personal Identification Number
PNP	Private Numbering Plan
PPS	Permanent Presentation Status
PSTN	Public Switched Telephone Network

PTT	Push To Talk
QoS	Quality of Service
RACF	Remote Activation of Call Forwarding
ROH	Receiver-Off-Hook
RST	Residential SIP Telephony
RTCP	Real-Time Control Protocol
RTP	Real-time Transport Protocol
RTT	Round Trip Time
SCA	Selective Call Acceptance or Shared Called Appearance
SCF	Selective Call Forwarding
SCR	Selective Call Rejection
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SLE	Screening List Editing
SMB	Small to Medium Business
SPP	Subscriber Programmable PIN
TDD	Telecommunications Device for the Deaf
UE	User Equipment
UI	User Interface
URI	Uniform Resource Identifier
URL	Uniform Resource Locator
VMWI	Visual Message Waiting Indicator
VoIP	Voice over Internet Protocol
VSC	Vertical Service Code
WIN	Wireless Intelligent Network
WS	Web Server

5 PACKETCABLE BUSINESS SIP SERVICES SCOPE

5.1 Overview

Increasingly, cable operators are interested in providing their customers with a wider range of endpoint devices. They include software-based phone clients, PDA's and feature-phones, mobile devices, and video-telephony terminals, all of which are typically connected to IP-based networks. Businesses are interested in blended voice and data applications such as unified messaging, web-access to feature management and configuration, and cross-platform alerts and messaging.

SIP is becoming the de-facto communication protocol for establishing voice and multimedia communications between endpoint devices on Internet Protocol networks. An increasing number of Customer Premises Equipment (CPE) and endpoint device manufacturers are building SIP-based voice products for business and enterprise environments. These devices encompass a wide range of telephone handsets and software clients that are compliant with standard SIP signaling. Many of them are capable of multimedia features such as data and video communications.

The leading business telephony application server vendors and IP-PBX manufacturers have also embraced SIP as a framework for voice and multimedia service delivery. As a result, much of the new feature development and service innovation that is occurring in enterprise and commercial telephony utilizes SIP-based products from leading equipment manufacturers.

The scope of the BSS effort is limited to specifying a multi-line voice service via a Centrex-like or hosted IP-PBX model. Non-facilities-based hosted solutions provide several benefits to business customer. Specifically, they are able to get advanced business features typically available only on enterprise PBX systems. Since the service is hosted by the cable operator, businesses save costs by eliminating management and operational expenses of running a PBX system.

Hosted service scenarios do not necessarily rely on embedded devices such as E-DVAs. In a hosted application, the endpoint devices may be IP telephone sets that are deployed on the customer Local Area Network (LAN). These client devices are capable of receiving call signaling and media directly from the MSO hosted telephony application servers or the MSO hosted IP-PBX system. A NAT and firewall enabled gateway device between the clients and the network can be an important component of a hosted solution.

This technical report describes the telephony call features and capabilities that approximate the services available to most PSTN business voice subscribers. Business SIP Services is an application built upon the PacketCable architecture described in [ARCH-FRM TR]. The PacketCable Architecture Framework technical report and related specifications specify the system components, the various functional groupings and the network interfaces necessary for delivery of services via a PacketCable network. The architecture (which is based on 3GPP IMS) leverages the use of triggers within the core network to determine when application level processing is required.

This report defines the Business SIP Services features from a user's point of view, without delving into many of the details of the network elements in the PacketCable network. Figure 1 illustrates the context and the interfaces that are within scope for this report.

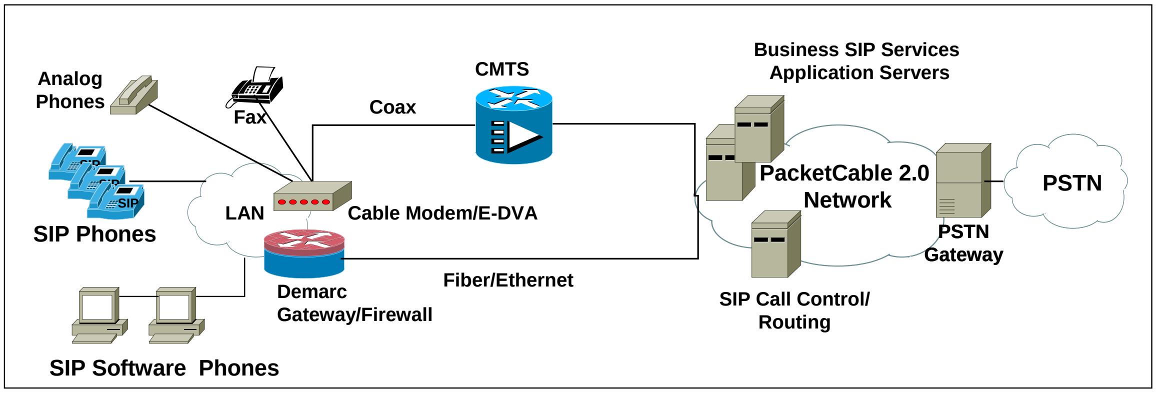

Figure 1 - Business SIP Services Context

This document defines requirements for the Business SIP Services capability set as they impact the following elements.

BSS User Equipment (UE): the functional element at the customer premise that attaches to the PacketCable network. UEs may take any number of forms, including clients embedded into cable modems, clients embedded into set-top boxes, clients embedded into SIP Phones, and software clients on PCs or laptops. UEs support several interfaces to the PacketCable network:

- Session Initiation Protocol (SIP): For signaling the creation, modification, and deletion of sessions.
- Media: For carrying audio or other media traffic, and related statistics.
- Network Address Translation (NAT) and Firewall Traversal: For permitting Business SIP Services features to operate through NAT and Firewall devices.
- Security: For securing interfaces.

BSS Application Server (AS): The Business SIP Services Application Server is a logical network component that implements some aspects of BSS features. The PacketCable architecture provides a framework for triggering SIP messages to the application server based on the service profile of a user.

5.2 Business SIP Services Feature Groupings

The Business SIP Services capabilities are divided into feature groups for easy reference. The feature groups are defined below:

- **Basic Calling Capabilities**: including call origination, call termination, addressing, quality of service, and basic call progress indicators.
- **Caller ID**: including all features associated with delivery and blocking of a caller's name and number.
- **Call Forwarding**: including all supported variants of the call forwarding feature.
- **Call Blocking**: including support for blocking specific categories of outbound and inbound calls.
- **Multi-Party Features**: including call waiting, hold, transfer, three-way calling and multi-party conference calling.
- **Call History**: including Auto Recall and Auto Callback.
- **Miscellaneous Features**: including do-not-disturb, distinctive alerting, message waiting indicators, speed dialing, and customer-originated call trace features.
- **Emergency Services**: including emergency calling and location identification capabilities.

- **Operator Services**: including busy-line verification and emergency operator interruption capabilities.
- **Lifeline Capabilities**: including capabilities for providing service during AC power outages.
- **Business-oriented Individual Line features**: including Automated Attendant, Direct Inward Dialing, Direct Outward Dialing, Extension Dialing, Alternate Number, Intercom, Shared Call Appearance, Sequential and Simultaneous Ring, Call Jump and Remote Office.
- **Business Oriented Group Line features**: including Account codes, Authorization codes, Hunt groups, Call Park features, Call Pickup features, Outbound Call Restriction, Charge Number service and Loudspeaker paging.
- **Web Portal services**
- **Third Party Call Control Features**
- **Email, Address Book and Directory Integration**

6 PACKETCABLE BUSINESS SIP SERVICES FEATURE DEFINITIONS

This section contains detail on each of the features that may be offered as part of PacketCable Business SIP Services. Service definitions are described for the simple case of a single UE emulating the legacy PSTN business voice service, and for the case where multiple UEs can be a part of the same Business SIP Services subscription.

The Business SIP Services suite of specifications is based on this technical report and consists of the following specifications.

- PacketCable Business SIP Services Feature Specification

  This specification defines specific requirements on the UE and Application Servers to implement Business SIP Services features. Some of the features are completely implemented on the UE; some are completely implemented on an Application server, while some others are implemented on both UE and Application Server. The MGC requirements for support of Business SIP Services features are specified in [CMSS1.5].

- PacketCable Business SIP Services Provisioning Specification

  This specification defines the requirements for provisioning a Business SIP Services UE.

6.1 Relation to RST Technical Report

The RST Technical Report [RST TR] documents the set of residential telephony features in a PacketCable network with SIP based UEs providing an analog user interface. Most of the residential features have equivalent versions in BSS.

[RST TR] has been designed for UEs with analog interfaces and requires that the UE implements behavior not typically found in UE implementations developed for non-Cable markets. This technical report allows alternative implementation of these features suitable for a broader range of UEs without requiring full behavior documented in [RST TR].

An RST-capable UE is expected to behave as documented in the [RST TR]. Additionally a BSS UE that contains an embedded Cable Modem and supports an analog interface is expected to behave as documented in [RST TR].

For BSS features which have a residential equivalent, it is assumed that the BSS AS is capable to interoperate with RST-capable UEs.

When it is mentioned in this report that [RST TR] applies, it is assumed that reference in [RST TR] to RST UE also mean BSS UE, unless noted otherwise in this report.

6.2 Basic Calling Capabilities

6.2.1 UEs and Public User Identities

One of the major differences between a traditional telephone network and a SIP Telephony network lies in the relationships between users, public user identities, UEs, and devices. A user may have multiple devices, each of which may host multiple UEs, and each UE may be registered to one or more public user identities. A public user identity can be a telephone number or it can be an alphanumeric identifier that makes sense in the context of a SIP Telephony service. Each public user identity is generally associated with a user. Multiple UEs may be beyond a single public user identity; in other words, a public user identity may be assigned to more than one UE. These are generally referred to as contacts (in SIP terms).

In SIP-based telephone networks, a single telephone number (public user identity) may be assigned to more than one UE. These UEs do not have to act as if they were multiple extensions on the same line, as in the traditional telephone network, where any telephone can be used to listen and speak in the same telephone conversation. Instead, UEs may act as independent telephones, each with their own line, such that a call to the telephone number causes all of the UEs assigned to that public user identity to ring, but the first one to pick up gets the call. There is requirement for BSS to emulate analog line extension behavior.

6.2.2 Basic Call for Multiple UEs per Public User Identity

The procedures for support of basic calls when a public user identity is simultaneously registered on multiple UEs (i.e., when the public user identity has multiple attachment points on the network), is as specified in [PKT 24.229].

6.2.3 Addressing

6.2.3.1 Dial String Entry

The Dial String Entry section of [RST TR] applies to BSS.

6.2.3.2 Initiating a Call or Feature via Dial String

The Initiating a Call or Feature via Dial String section of [RST TR]applies to BSS.

6.2.3.3 Creating a Routable Address from a Dial String

The Creating a Routable Address from Dial String section of [RST TR] applies to BSS.

6.2.4 Call Progress Indications

The Call Progress Indications section of [RST TR]applies to BSS.

6.2.5 Announcements

The Announcements section of [RST TR]applies to BSS.

6.2.6 Permanent Sequence and Lockout

The Permanent Sequence and Lockout section of [RST TR] applies to BSS.

6.2.7 No Answer Timeout

The No Answer Timeout section of [RST TR]applies to BSS.

6.2.8 UE Status

The UE Status section of [RST TR] applies to BSS, with the exception that a call placed to a public user identity for which there are no registered UEs would trigger the BSS feature "Call Forwarding - Endpoint/Network not Available".

If a call is placed to a public user identity for which there are no registered UEs, then the public user identity is considered unreachable, not busy. An unreachable UE has similar semantics to a mobile phone that is out of range of a mobile phone provider's network. This can be an important semantic distinction that will be used to trigger the service "Call Forwarding - Endpoint/Network not Available".

A UE that is in a lockout state is considered to be busy, and should respond to new calls accordingly.

6.2.9 Quality of Service

In addition to what is documented in [RST TR], the following apply to BSS.

PacketCable Residential SIP Telephony Feature specification [RSTF] mandates the UE to mark the upstream IP packets with configurable DSCP value, based on the IP packet content, whether it is Non-Emergency Media, Emergency Media, Non-Emergency Signaling, or Emergency Signaling. CMTS is responsible for policing the DSCP value for upstream IP packets. This mechanism is anticipated to be re-used for BSS. It is assumed that any routers or other equipment between UE and CM will not alter the DSCP value set by the UE.

The QoS in the Customer Premises LAN network is out of scope of BSS.

6.2.10 Advanced User Interfaces on SIP Phone

Business SIP Services are not designed targeted at a specific device type or assuming any specific user interface capability. The user client to network signaling (SIP, XCAP etc.) is also device-agnostic, it conveys sufficient information to enable service specific requirements including state changes at both the network and user maintained service state machines where applicable. It is therefore incumbent upon the specific device type enabling user service to render to the user sufficient information to enable a meaningful and coherent user interaction with the Business SIP service set. Following is additional discussion as to the areas of consideration of Business SIP devices with regard to differing user interface capabilities.

User interactions traditionally defined within Telcordia and other specifications are predicated on the assumption that the user device is a 'black phone'. That is an analog device with a limited user interface (traditional keypad and hook-switch flash) and no mechanism to convey feedback with regard to service or operational state outside of dial tone and other audio indication application.

The advent of SIP phones and soft-clients enables a class of Business SIP capable devices that have rich UI capabilities or in the cases of soft-clients may not even apply dial-tone. In such cases service specific requirements that pertain to the application of specialized dial tone cadences or specific sequences involving hook-switch flash or application of audio and so forth may not be applicable.

Thus non-analog devices supporting Business SIP services need to provide appropriate mechanisms and indications based on the capabilities of the device in the following service scenarios:

- Feature/Service State Indications.

  In analog clients these are typically realized via the application of unique dial tones or potentially ring-splash in the case of a Call Forwarding Variable (CFV). A device with a different UI or richer UI may make use of alternative mechanisms such as lamps, icons, dialog boxes and so forth.

- Feature Activation.

  In analog clients this typically requires the entering of a Vertical Service Code (VSC) coupled with potentially the use of hook-switch flash. A device with a richer user interface may make use of alternative mechanisms such as assigned feature keys, soft keys and so forth.

6.3 Caller ID

The Caller ID section of [RST TR] applies to BSS.

6.3.1 Calling Number Delivery and Calling Name Delivery

Calling Number Delivery (CND) provides customer premises equipment (CPE) with the date, time and directory number of an incoming call. Calling Name Delivery (CNAM) provides (CPE) with the date, time and calling party's name of an incoming call.

Calling Name and Number Delivery functionality is described in Telcordia documents [GR 31] (CND), [GR 1188] (CNAMD), and [GR 30] (Delivery of data).

The user interface descriptions in [GR 31] are followed for UEs with analog interfaces. UEs with other interfaces are allowed to provide alternative user interfaces because different types of UEs have widely varying user interface capabilities.

Calling Name and Number is also delivered with calls that are routed to voice mail systems.

6.3.1.1 Feature Interactions

Interactions with other features are as specified in [GR 31] and [GR 1188], with limitations as follows:

- CND subsection of [GR 1188] applies except that usage sensitive CND and CNAMD do not apply.
- For interactions with AC and AR, see Section 6.7.

6.3.2 Caller ID Per-Line Blocking

Caller ID Per-Line Blocking section of [RST TR] applies to BSS.

6.3.2.1 Feature Interactions

Interactions with other features defined in this specification are as specified in [GR 391], with limitations as follows:

- [GR 391] describes multiple features. Only the interactions pertaining to per-line name and number blocking apply here.
- For Business SIP, there is only one PPS (Permanent Presentation Status) value governing the presentation of both name and number. References in [GR 391] to the PPS for either name or number refer to this single value.

6.3.3 Caller ID Per-Call Blocking and Delivery

CIDS Suppression is defined by [GR 391]. To use this feature the caller does the following:

- As part of the actions required to initiate a call, enters the assigned vertical feature code. The feature code may be provisioned in the UE as part of a digit map. (The suggested default value is *67.)
- May receive dial tone, or a confirmation tone followed by a recall dial tone or some other indication appropriate to the device that indicates that second stage dialing should commence.
- Either on reception of appropriate progress indication or as part of the initial dial string, dials the telephone number of the called party.

For this one call, the Permanent Presentation Status value is overridden with the value "anonymous". If the caller abandons the call attempt at any point prior to completion of dialing the called party number, the feature has no effect.

The CIDS Delivery feature is defined by [GR 391]. To use this feature the caller does the following:

- As part of the actions required to initiate a call, enters the assigned vertical feature code. The feature code may be provisioned in the UE as part of a digit map. (The suggested default value is *82.)

- May receive dial tone, or a confirmation tone followed by a recall dial tone or some other indication appropriate to the device that indicates that second stage dialing should commence.
- Either on reception of appropriate progress indication or as part of the initial dial string, dials the telephone number of the called party.

For this one call, the permanent presentation status value is overridden with the value "public". If the caller abandons the call attempt at any point prior to completion of dialing the called party number, the feature has no effect.

6.3.3.1 Feature Interactions (CIDS-D)

AC

AC allows customers to place calls to the last call they called. CIDS-S or CIDS-D can be activated before AC activations and reactivations as well as before the original call is made. When an AC call is finally established, the most recent CIDS-x that the AC customer activates determines the presentation status associated with the AC customer.

If no CIDS-x is invoked immediately before AC activation or reactivation, then the AC is invoked using the same presentation status that was used when the customer originally entered the telephone number to place the call.

If the feature code for CIDS-S is entered, and followed by the feature code for AC, then the AC is invoked with a presentation status of "anonymous". This applies to both activation and reactivation of AC.

AR

Auto Recall allows customers to place calls to the last call they received. CIDS-S or CIDS-D can be activated before AR activations and reactivations. When an AR call is finally established, the most recent CIDS-x that the AR customer activates determines the presentation status associated with the AR customer.

If the feature code for CIDS-S is entered, and followed by the feature code for AR, then the AR is invoked with a presentation status of "anonymous". This applies to both activation and reactivation of AR.

An AR customer may use this feature on the last call that the customer received regardless of the caller's presentation statuses. However, if the call that the AR customer received was originally marked "anonymous", the network does not disclose the Caller ID to the AR customer when the call is finally established and the AR customer gets ring back.

Calling Number Delivery

If a caller activates CIDS-D then the called party will receive Caller ID, including calling number, regardless of the permanent presentation status. Detailed results are covered in Section 6.3.

Calling Name Delivery

If a caller activates CIDS-D then the called party will receive Caller ID, including calling name, regardless of the permanent presentation status. Detailed results are covered in Section 6.3.

Operator Services

A caller should be able to access CIDS-x and operator services for the same call. A caller is able to dial the CIDS-x feature access code before dialing a number that invokes operator services. The resulting presentation status is used for the call.

Screening List Services

If the called party invokes screening list editing procedures to add the last calling number to a screening list and the caller's presentation status is anonymous, an "anonymous" indication is included in the new screening list entry. However, an "anonymous" entry in the screening list is displayed or otherwise be made available to the customer.

Speed Dialing

A CIDS-x feature activation code could be entered as part of an entry in a speed-calling list.

When speed dialing is invoked, if the selected speed-calling list entry contains a CIDS-x feature activation code, then the call is placed with that feature activated.

A speed-calling invocation could be entered following the entry of the CIDS-x feature activation code. In this case the call is placed with that feature activated.

It is an error to enter a CIDS-x feature activation code followed by a speed-calling invocation if the selected speed-calling list entry contains a CIDS-x feature activation code.

Three-Way Calling

A three-way call has two legs. The originator of each leg of the call can use CIDS-x during the initiation of that leg.

6.3.3.2 Feature Interactions (CIDS-S)

Anonymous Call Rejection (ACR)

Activating CIDS-S results in call rejection if the called party has invoked the ACR feature.

Automatic Callback (AC)

AC allows customers to place calls to the last call they called. CIDS-S or CIDS-D can be activated before AC activations and reactivations as well as before the original call is made. When an AC call is finally established, the most recent CIDS-x that the AC customer activates determines the presentation status associated with the AC customer.

If no CIDS-x is invoked immediately before AC activation or reactivation, then the AC must be invoked using the same presentation status that was used when the customer originally entered the telephone number to place the call.

If the feature code for CIDS-S is entered, and followed by the feature code for AC, then the AC must be invoked with a presentation status of "anonymous". This applies to both activation and reactivation of AC.

Automatic Recall (AR)

AR allows customers to place calls to the last call they received. CIDS-S or CIDS-D can be activated before AR activations and reactivations. When an AR call is finally established, the most recent CIDS-x that the AR customer activates determines the presentation status associated with the AR customer.

If the feature code for CIDS-S is entered, and followed by the feature code for AR, then the AR must be invoked with a presentation status of "anonymous". This applies to both activation and reactivation of AR.

An AR customer may use this feature on the last call that the customer received regardless of the caller's presentation statuses. However, if the call that the AR customer received was originally marked "anonymous", the SPCS shall not disclose the Caller ID to the AR customer when the call is finally established and the AR customer gets ring back.

Calling Number Delivery

If a caller activates CIDS-S then the called party will not receive Caller ID, including calling number.

Calling Name Delivery

If a caller activates CIDS-S then the called party will not receive Caller ID, including calling name.

Operator Services

A caller should be able to access CIDS-x and operator services for the same call. A caller is able to dial the CIDS-x feature access code before dialing a number that invokes operator services. The resulting presentation status is used for any call the operator makes on behalf of the caller.

Screening List Services

If the called party invokes screening list editing procedures to add the last calling number to a screening list and the caller's presentation status was anonymous, an "anonymous" indication is included in the new screening list entry. However, an "anonymous" entry in the screening list will not be displayed or otherwise be made available to the customer.

Speed Dialing

A CIDS-x feature activation code may be entered as part of an entry in a speed-calling list.

When speed dialing is invoked, if the selected speed-calling list entry contains a CIDS-x feature activation code, then the call is placed with that feature activated.

A speed-calling invocation may be entered following the entry of the CIDS-x feature activation code. In this case the call is placed with that feature activated.

It is an error to enter a CIDS-x feature activation code followed by a speed-calling invocation if the selected speed-calling list entry contains a CIDS-x feature activation code.

Three-Way Calling

A three-way call has two legs. The originator of each leg of the call may use CIDS-x during the initiation of that leg.

6.3.4 Caller ID with Call Waiting

Caller ID with Call Waiting section of [RST TR] applies to BSS.

6.3.5 Anonymous Call Rejection (ACR)

Anonymous Call Rejection (ACR) section of [RST TR] applies to BSS.

6.3.5.1 Feature Interactions

The ACR service has the following feature interactions.

Call Waiting

If the called user has activated ACR, then ACR takes precedence over the Call Waiting service.

If ACR is activated while a call is waiting, the state of the waiting call is not changed.

Caller ID Per-Call Blocking

If the activation of Caller-ID Per-Call blocking by an originating user results in a presentation status of anonymous for the call, and that user is not an authorized user with ACR override, then all calls from that user are rejected by a subscriber with ACR.

Caller ID Per-Line Blocking

If the originating user has subscribed to Caller-ID Per-Line blocking and that user is not an authorized user with ACR override, then all calls from that user are rejected by a subscriber with ACR.

Call Forwarding

If the served user has activated ACR, then ACR takes precedence over Call Forwarding Variable.

If the served user has activated ACR, then ACR takes precedence over Call Forwarding Busy.

If the served user has activated ACR, then Selective Call Forwarding (SCF) takes precedence over ACR. That is, if the restricted identity matches that of an entry in the subscriber's SCF screening list, then the call is routed to the destination specified by SCF and ACR is not applied.

Automatic Callback (AC)

If the served user has activated ACR, then ACR takes precedence over AC.

If AC has been requested and the AC target activates ACR during the recall period, then the call attempt based on the AC service is subject to ACR.

Automatic Recall (AR)

If the served user has activated ACR, then ACR takes precedence over AR.

Emergency Services

Emergency call back overrides Anonymous Call Rejection.

Operator Services Emergency Interrupt overrides Anonymous Call Rejection.

Do Not Disturb

Do Not Disturb takes precedence over Anonymous Call Rejection.

6.4 Call Forwarding

The Call Forwarding section of [RST TR] applies to BSS, with the exception that BSS adds one more Call Forwarding flavor, called "Call Forwarding Endpoint/Network Not Available".

6.4.1 Call Forwarding Variable (CFV)

The Call Forwarding Variable section of [RST TR]applies to BSS.

6.4.1.1 Feature Interactions

Call Forwarding Variable has no interaction with calls that originated from a public identity with call forwarding activated.

Call Blocking

The BSS AS does not allow the activation of the call forwarding variable to blocked numbers. The CFV provides an error notification in response to this activation request.

Call Forwarding Busy Line (CFBL)

When CFV is deactivated, CFBL takes precedence.

When activated, CFV takes precedence over CFBL.

Call Waiting (CW)

On a line that is activating CFV, CW is not applied after the answer is received or the confirmation tone is returned.

Cancel Call Waiting (CCW)

When activated, CFV takes precedence over CW. Although CCW has no effect in this situation, the UE still returns recall dial tone after the customer dials the CCW access code and receive the telephone number dialed by the customer. The UE typically allows a customer to disable CW while CFV is activated.

The UE would typically allow customers who subscribe to both CFV and CW to dial the access codes for both on a single call as long as the CCW access code is dialed first. For example, the following dialing sequence is allowed:

1. Customer dials *70 (CCW)
2. Recall dial tone returned
3. Customer dials *72 (CFV)
4. Return recall dial tone
5. Customer dials the telephone number of the remote station
6. Remote station is rung
7. Remote station answers

This sequence results in CW being disabled for the duration of the call, and CFV being activated to the remote station.

IDDD Via an Operator System

The BSS AS must not allow 010 and 01+ as "forward to" numbers.

Call forwarding to international destinations outside World Zone 1 on an 011+ basis is permitted, but not required, when these inter-LATA calls utilize Inter-LATA Carrier/International Carrier (IC/INC) interconnection (Feature Group D) access arrangements. If call forwarding to IDDD is allowed on an 011+ basis, the telephone company should be provided the ability to inhibit lines from activating call forwarding to 011+ destinations on an office basis and, optionally, on a per line basis.

Emergency Service (911)

The BSS AS does not allow 911 as a "forward to" number.

IC/INC Interconnection

A 10XXX (+1) + 7/10 digit number may be the remote number whether the CFV feature is activated by the customer or by subscriber provider provisioning. Calls terminating from an IC/INC may be forwarded. When call forwarding is initiated without a 10XXX prefix, calls should be forwarded via the presubscribed carrier.

The remote number should be a 7- or 10-digit number, but call forwarding can also be initiated by means of AD2 by dialing the "call forwarding activation code" +10XXX + a "speed dialing number."

The BSS AS does not allow 950-WXXX as a "forward to" number.

Call forwarding to international destinations outside World Zone 1 on a 011+ basis is permitted but not required when these inter-LATA calls utilize IC/INC interconnection (FGD) access arrangements. If call forwarding to IDDD is allowed on an 011+ basis, the telephone company should be provided the ability to inhibit lines from activating call forwarding to 011+ destinations on an office basis and, optionally, on a per line basis.

A CFV customer activates the feature in two ways. In one way, the customer follows the activation procedure and the attempted call is answered, activating the feature. In the second way, the call is not answered. However, if the entire activation procedure is repeated within two minutes to the same DN, the feature is activated. In this second method, the customer need not indicate the same carrier in both activation attempts (using 10XXX prefix or pre-subscription). The carrier given on the second attempt is the one used for forwarded calls.

Error conditions during activation and during an attempted forwarding of a call are handled as described in [GR 580].

Do Not Disturb

Do Not Disturb takes precedence over Call Forwarding Variable.

Operator Services

The CFV does not allow 0-, 0+, or X11 as a "forward to" number.

For an exception, see the interaction of call forwarding variable with IDDD via an operator system.

The BSS AS would typically return a 486 to a Busy Line Verify INVITE with Join header.

Selective Call Forwarding

Selective Call Forwarding takes precedence over Call Forwarding Variable

6.4.2 Call Forwarding Don't Answer (CFDA)

Call Forwarding Don’t Answer (CFDA) section of [RST TR] applies to BSS.

6.4.2.1 Feature Interactions

CFDA has no interaction with calls originated from a public identity with call forwarding activated.

Call Forward Variable (CFV)

CFV takes precedence over CFDA.

Call Forwarding Endpoint/Network Not Available (CFNA)

When the UE is not registered on the network, CFNA takes precedence over CFDA for determination of the forward-to address.

Call Waiting (CW)

If a CFDA subscriber receives an additional call offering via CW and the subscriber does not answer the additional call, CFDA is applied upon expiry of the no answer timer.

Call Blocking

The selection of the remote address is not allowed to violate outbound call blocking restrictions.

Do Not Disturb

Do Not Disturb takes precedence over CFDA.

Selective Call Forwarding

Selective Call Forwarding takes precedence over CFDA.

6.4.3 Call Forwarding on Busy Line (CFBL)

The Call Forwarding on Busy Line (CFBL) section of [RST TR] applies to BSS.

6.4.3.1 Feature Interactions

Call Forwarding Busy Line has no interaction with calls originating from a public identity with call forwarding activated.

Call Forward Variable (CFV)

CFV takes precedence over CFBL.

Call Waiting (CW)

If a CFBL subscriber is also subscribed to CW, then the UE invokes the Call Waiting procedures, as defined in Section 6.6.1, for the first additional call offered. Further, if the UE is handling two calls under CW, the UE returns a 486 Busy response to subsequent INVITE requests.

Call Blocking

The remote address is subject to outbound call blocking restrictions.

Operator Services

Busy Line Verification takes precedence over CFBL.

Emergency Interrupt takes precedence over CFBL.

Do Not Disturb (DND)

For calls to CFBL subscribers who have DND activated, the UE rejects all INVITE requests with a 486 Busy response.

6.4.4 Selective Call Forwarding (SCF)

The Selective Call Forwarding (SCF) section of [RST TR] applies to BSS.

6.4.4.1 Feature Interactions

Selected Call Forwarding has no interaction with calls originated from a public identity with call forwarding activated.

Call Blocking

The remote address for SCF is subject to outbound call blocking restrictions.

Speed Dialing

Speed dialing codes may be used to specify the SCF remote DN.

IC/INC Interconnection

The BSS AS does not allow forward-to addresses to be 950-WXXX numbers.

When a call is forwarded using an Inter-exchange or International Carrier (IC/INC), the BSS AS provides the address of the forwarding UE to the carrier for billing purposes.

Call Forwarding

SCF always takes precedence over other Call Forwarding services assigned to the same UE (Call Forwarding Variable, Call Forwarding Don't Answer, and Call Forwarding Busy Line). When SCF and another Call Forwarding service exists on the same UE, the remote addresses specified for the respective services may differ and are expected to differ.

If a forwarded call is received at a line that has SCF active, the BSS AS matches the Screen List against the originating address (not a forwarding address). If the originating address is indicated on the screening list, the BSS AS forwards the call. See above for restrictions concerning multiple call forwarding sequences.

Distinctive Ringing/Call Waiting (DRCW)

SCF takes precedence over Distinctive Ringing/Call Waiting.

Do Not Disturb (DND)

SCF takes precedence over Do Not Disturb.

Call Waiting (CW)

SCF takes precedence over Call Waiting.

Automatic Callback (AC)

If a person attempts to activate AC to a UE that has SCF active, the AC customer's address is checked against the SCF list. If the address is on the list, BSS AS rejects the AC request since the status of the remote line cannot be checked. If the number does not match the list, the BSS AS continues to execute SCF.

Automatic Recall (AR)

If a person attempts to activate AR to a line that has SCF active, the AR customer's address is checked against the SCF list. If the address is on the list, the BSS AS rejects the AR request since the status of the remote line cannot be checked. If the number does not match the list, the BSS AS continues to execute SCF.

6.4.5 Remote Activation of Call Forwarding

The Remote Activation of Call Forwarding section of [RST TR] applies to BSS.

Furthermore, a UE with non-analog interfaces can provide alternative user interfaces because different types of UEs have widely varying user interface capabilities. In addition to the wide range of UEs, an operator may offer a web client mechanism which allows a subscriber to manage his or her service data.

6.4.5.1 Feature Interactions

Call Forward Variable

RACF is simply another way to program CFV, so RACF and CFV share the same forward-to number and activation status.

The RACF subscriber is also subscribed to CFV.

If the RACF PIN code is to be user programmable, the RACF subscriber is subscribed to SPP.

Restrictions on the forward-to address are identical to those for CFV.

6.4.6 Call Forwarding to Voice Mail

The Call Forwarding to Voice Mail section of [RST TR] applies to BSS. In addition, CFNA may also be a trigger to forward a call to voice mail. Call Forwarding Endpoint/Network Not Available (CFNA).

Call forwarding Endpoint/Network Not Available (CFNA) is a feature that allows forwarding of all calls to the subscriber's public identity to another location if the UE is not registered, or for some reason network problems prevent the call from being delivered to the UE. Detection of network problems by the BSS AS may be done based on signaling or other means not specified in the specification.

If the UE is not registered, CFNA condition takes precedence over CFV.

Once call forwarding programming is successful, the forwarded-to number is stored in such a manner that it remains persistent even in the case of complete power loss in the AS.

The activation and deactivation of the CFNA feature is performed by the service provider as part of the service provisioning for a user. The associated provisioning procedures are outside of the scope for this specification. However, the feature activation/deactivation status is made available to the BSS AS.

If a public user identity has CFNA enabled and the incoming call receives a response of "not found" or "temporarily unavailable", the incoming call should be forwarded to the forward-to address.

6.4.6.1 Feature Interactions

CFDA has no interaction with calls originated from a public identity with call forwarding activated.

Call Forwarding Variable (CFV)

CFV takes precedence over CFNA.

Call Blocking

The selection of the remote address is not allowed to violate outbound call blocking restrictions.

6.5 Call Blocking

The Call Blocking section of [RST TR] applies to BSS.

6.5.1 Outbound Call Blocking

The Outbound Call Blocking section of [RST TR] applies to BSS.

6.5.1.1 Feature Interactions

Speed Dialing and Call Forwarding

Speed Dialing and Call Forwarding are invoked prior to the OCB feature.

6.5.2 Collect Call Blocking

This is a network only-based feature relying on LIDB (Line Identification Database) for feature status.

The Collect Call Blocking section of [RST TR] applies to BSS.

6.5.2.1 Feature Interactions

Collect Call Blocking occurs before the call reaches the PacketCable network; hence there are no feature interactions.

6.5.3 Selective Call Acceptance (White List)

The Selective Call Acceptance feature enables a user to define criteria that causes certain incoming calls to be allowed. If an incoming call meets user-specified criteria, the call is allowed to complete to the user. All other calls are blocked and the caller is informed that the user does not wish to receive the call. The user controls the service via a web interface or via telephone handset. This provides the ability to establish the criteria sets for determining which calls are allowed to complete. A criteria set is based on incoming calling line identity, time of day, and day of week. Multiple criteria sets can be defined.

6.5.3.1 Feature Interactions

The Selective Call Acceptance feature interaction is as follows.

Basic Call

Selective Call Acceptance takes precedence over basic calling functionality for call termination.

Call Forwarding Variable

Selective Call Acceptance takes precedence over Call Forwarding Variable.

Call Forwarding Don't Answer

Selective Call Acceptance takes precedence over CFDA.

Call Waiting

Selective Call Acceptance takes precedence over Call Waiting.

Do Not Disturb

Do Not Disturb takes precedence over Selective Call Acceptance.

6.5.4 Selective Call Rejection (Black List)

The Selective Call Rejection feature enables a user to define criteria that cause certain incoming calls to be blocked. If an incoming call meets user-specified criteria, the call is blocked and the caller is informed that the user is not accepting calls. The user controls the service either via telephone handset or via a web interface. This provides the ability to establish the criteria sets for determining which calls require blocking. A criteria set is based on incoming calling line identity, time of day, and day of week. Multiple criteria sets can be defined.

Except where noted, this feature conforms to [GR 218].

6.5.4.1 Feature Interactions

The Selective Call Rejection feature interaction is as follows.

Basic Call

Selective Call Rejection takes precedence over basic calling functionality for call termination.

Call Forwarding Variable

Selective Call Rejection takes precedence over Call Forwarding Variable.

Call Forwarding Don't Answer

Selective Call Rejection takes precedence over CFDA.

Call Waiting

Selective Call Rejection takes precedence over Call Waiting.

Do Not Disturb

Do Not disturb takes precedence over Selective Call Rejection.

6.6 Multi-Party Features

6.6.1 Call Waiting

The Call Waiting section of [RST TR] applies to BSS, with the following additions/clarifications.

With the Call Waiting feature activated on the UE, the network will offer any incoming call to the UE; it is up to the UE to decide if or not the call should be held as a waiting call.

Call Waiting feature for UEs without analog interface can be deactivated in a manner consistent with the user interface of these UEs.

6.6.1.1 Feature Interactions

The call waiting feature has the following interactions.

Basic Call

The UE responds with a 486 Busy Here final response when a new INVITE is received and the UE is not yet in a stable two-way call (for example, while dialing). Further, the UE responds with a 486 Busy Here final response when a new INVITE is received and when the UE is in the process of tearing down the call but has not yet sent a BYE.

Distinctive Alerting

Distinctive Alerting may be applied for a call in wait, as per Section 6.6.1.

Call Transfer

The precedence of call waiting with call transfer depends on the state of the transfer attempt.

When the subscriber executes a hook flash or other equivalent action while a call is waiting, the UE answers the waiting call instead of performing a call transfer.

Call waiting is disabled after initiating call transfer. The UE rejects an incoming call with a 486 Busy Here final response while the call transfer feature executes.

Three-way Call

Call waiting takes precedence over a three-way call when the three-way call is not executing. Call waiting is disabled after initiation of a three-way call depending on the resource availability of the UE.

Emergency Call Origination

The UE must not execute the Call waiting feature when the existing call is an emergency call.

6.6.2 Call Hold

The Hold section of [RST TR] applies to BSS.

6.6.2.1 Feature Interactions

Basic Call

For a UE with analog interface, if the UE already has a call on hold when it receives a new INVITE, it rejects the incoming call with a 486 Busy Here response.

Transfer Call

The UE is able to transfer the non-held call without impacting the held call.

Conference Call

The UE does not permit a conference call between the non-held and held calls.

The UE permits conference calls between non-held parties.

Call Waiting

For a UE with analog interface, if the UE has a held call and an active call at the same time, it does not provide call waiting, because hook flash is used to retrieve the held call.

Emergency Calling

A UE does not place a call on hold, for the case of a call that it has recognized as an emergency call.

6.6.3 Music On Hold

The Music on Hold feature uses pre-recorded audio to fill the silence that would otherwise be heard by callers placed on hold. Although music is commonly used, the audio stream could provide information considered of value to the caller, such as business hours, products, services and promotions.

A Business Group Administrator is typically able to control the audio to be played to callers, for example using a Web portal or their phone. This could involve selecting pre-recorded music from a library provided by the service provider, recording a custom message or uploading custom audio files. The administrator may also specify which lines have the feature enabled, for example only playing music when calls to the company switchboard are held.

When a user connected to a conference bridge places their call on hold, it is undesirable that music is introduced into the conference. One solution is to make the above example the default behavior; by only playing music when calls incoming to the company are held this default behavior would ensure that music on hold is not played to participants connected to a meet-me conference bridge when one of the caller places his call on hold. With SIP-based conference bridges, it may be possible to detect that the call is connected to a bridge and suppress the playing of music. Other solutions require that the feature is not enabled on lines typically used to dial into conference bridges, or require that the user remembers to manually disable Music on Hold on a temporary basis.

6.6.3.1 Feature Interactions

Multi-party conference

If the BSS AS is aware that the UE is involved in a multi-party conference, music on hold is not played.

6.6.4 Consultation Hold

Consultation Hold feature enables a user to put the caller on hold and make a consultation call to another party. To initiate consultation hold, the user presses the flash hook on an analog phone (or taking an appropriate user-interface-specific action) and 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 a second time on an analog phone (or taking an appropriate user-interface-specific action).

6.6.4.1 Feature Interactions

3WC and Multi-Party Conference

For UEs with analog interface support, the 3WC and Multi-Party Conference Calling take precedence over the Consultation Hold feature. Therefore, if either of the 3WC or the Multi-Party Conference Calling is activated, the user will conference in (rather than drop) the add-on party after the second hook flash. For the UEs that support other type of user interfaces, this interaction may not apply.

6.6.5 Call Transfer (CT)

The Call Transfer section of [RST TR] applies to BSS.

6.6.5.1 Blind Call Transfer (BCT)

The Blind Call Transfer (BCT) section of [RST TR] applies to BSS, with the following addition.

The transferor, transferee, and transfer-to parties may be in the same Business Group, different Business Groups, or even in different networks (e.g., where the transferor is an IP-Centrex user while the transferee and transfer-to users are PSTN-based users).

6.6.5.2 Consultative Call Transfer (CCT)

The Consultative Call Transfer section of [RST TR] applies to BSS.

6.6.5.3 Call Transfer Restrictions

The Call Transfer Restrictions section of [RST TR] applies to BSS.

6.6.5.4 Feature Interactions

Basic Call

The transferor's UE uses the same data as basic call (see Section 6.1) for the second call origination.

Call Waiting

If the subscriber's UE is engaged in an active call, a new INVITE is received at the subscriber's UE, and the subscriber has the Call Waiting feature, then, for as long as the UE is engaged in two dialogs, the subscriber's UE does not invoke the Call Transfer feature when the subscriber provides a hook flashes or other equivalent indication. For analog clients only, if the subscriber's UE receives a new INVITE at any point after the subscriber has initiated a transfer, and the subscriber has the Call Waiting feature, then the subscriber's UE responds to the new INVITE with a 486 Busy Here response.

Three-Way Calling

Call Transfer takes precedence over three-way calling.

Call Blocking, Outbound (In General)

If the subscriber invokes the CT feature and dials a blocked phone number in response to the second call leg prompt, the BSS AS blocks the call according to the procedures in Section 6.5.

Emergency Calling

When the subscriber's UE is engaged in an active Emergency Call and the subscriber attempts to initiate a second consult call leg, the subscriber's UE rejects this attempt and continues with the Emergency Call instead.

Caller ID Blocking

When sending a REFER, if caller-ID blocking is enabled for the transferor's public identity, the transferor's UE includes an anonymized Referred-By header or no Referred-By header as defined in [RFC 3892].

6.6.6 Three-Way Calling (3WC)

The Three Way Calling (3WC) section of [RST TR] applies to BSS. Furthermore, the following clarifications apply as well.

To drop the third party from the established 3WC session and remain connected with the second party, the subscriber presses the hook flash (or a button appropriate for the UE’s user interface) again.

The 3WC feature is implemented on the UE or with the support of a network-based conference bridge, dependent on the service provider preference.

6.6.6.1 Feature Interactions

Basic Call

For locally bridged calls, once a controller's UE is in a 3WC and detects a hook flash or other feature stimulus, it does not give the user the ability to initiate another basic call. If the subscriber is also Multi-Party Conferencing capable it is possible for additional calls to be supported by transferring the existing 3WC to a network hosted conference bridge; thus freeing up resources on the UE. For UEs however that rely on hook flash as a feature stimulus and have no other UI or stimulus capabilities this is problematic as the act of a hook flash within an existing 3WC results in the take down of the second established call leg and the reversion of the call to two-party.

A non-controller's UE has no knowledge that he or she is in a 3WC, so the non-controller UE may put the 3WC on hold if it detects a hook flash or other stimulus, and allow the user to make a new outgoing call.

Call Waiting

Call waiting takes precedence over a three-way call when the three-way call is not executing. Call waiting is disabled after initiation of a three-way call.

Call Transfer

When a controlling UE is in a 3WC, it cannot perform Call Transfer by a hook flash (if supported), as this disconnects the third party in the 3WC.

A controlling UE of a 3WC that has the Call Transfer feature enabled transfers when it detects on-hook or other call takedown condition.

A non-controlling UE has no knowledge that it is in a 3WC, so it can perform a call transfer.

Call Hold

Call hold takes precedence over three-way call.

Operator Service

When a controlling UE is in a 3WC, and a Busy Line Verify is received, the controlling UE may reject the INVITE if resources are not available to join the operator in the 3WC.

Emergency Call

A UE is forbidden to place an emergency call on hold when it detects a hook flash or other equivalent stimulus. When a party is in a two-party call, and the 3WC feature is enabled, the UE puts the first party on hold when it detects a hook flash or other equivalent stimulus and then it begins digit collection. The UE establishes a session with the PSAP if an emergency call number is dialed. After the emergency call is answered, the UE may join the emergency operator and the held party to form a 3WC if it detects a hook flash or other equivalent stimulus. If the UE detects a call take down action (e.g., on hook) before the emergency call is answered, the UE transfers the call. Once the 3WC is established, the UE ignores any detected hook flashes or other stimulus action. While in a 3WC with an emergency operator, the UE adheres to the emergency call procedures.

6.6.7 Multi-Party Conference Calling

The Multi-Party Conference Calling feature is similar to 3WC, with the difference that it allows the user to have more than two other parties to the conference call. The maximum number of parties allowed in the conference call is configurable by the operator; it depends on the capability of the network-based conference bridge employed by the operator for the implementation of this feature.

6.7 Call History Features

6.7.1 Auto Recall (Last Call Return)

The Auto Recall (Last Call Return) section of [RST TR] applies to BSS, with the following additions.

Multiple UEs with the same public user identity may be registered. In this case, if one of the UEs executes AR, then that UE uses the calling address received at that UE if the last received call was not anonymous. If the last call received was anonymous, the AR target address is the address (network stored) of the caller associated with the last received call at this UE.

6.7.1.1 Feature Interactions

The feature interactions of Automatic Recall feature are specified in [GR 227].

Calling Name Delivery

When AR is used with the CNAM feature, the called station's name need not be delivered to the customer when special ringing is applied. This differs from the requirements of [GR 227].

Caller ID Blocking

An AR request to a target which previously withheld their number is possible; details of the anonymous target are not revealed to the AR requestor.

Call Forwarding

An AR request to a target with Call Forwarding (see Section 6.4) active is should be rejected instead of being forwarded. The call to the line which originated the AR request (with the special ringing pattern) rings the line without being forwarded.

Call Rejection

A call which is rejected is not classified as the last incoming call.

An AR request to a target with a service which rejects calls (e.g., Anonymous Call Rejection, Do Not Disturb) is rejected if a normal call from the subscriber to the target would be rejected. The call to the line which originated the AR request (with the special ringing pattern) is not rejected by any Call Rejection service.

Auto Callback

Concurrent AR and AC requests from a user to the same target may be treated as one request.

Simultaneous/Sequential Ring

If the AR target is a Simultaneous or Sequential Ring subscriber, busy/idle monitoring only occurs on the subscriber's own line and not any of the other Simultaneous/Sequential ring destinations.

Account/Authorization Codes

The call type assigned to AR calls will typically have the same classification as a normal call to the target and so may require an Account/Authorization code to be provided.

Hunt Groups

Multiple activations from different lines in a Hunt Group to the same target are allowed and are treated as separate AR requests. The call to the line which originated the AR request (with the special ringing pattern) does not alert other lines in the hunt group. It is an implementation choice whether AR requests which target a Hunt Group are accepted or rejected.

Call Pickup

The call to the line which originated the AR request (with the special ringing pattern) cannot be picked up from another line using Call Pickup. The call to the AR target can be picked up.

Outbound Call Blocking/Call Restriction

The call type assigned to AR calls will typically have the same classification as a normal call to the target and so may be denied by the Outbound Call Blocking/Restriction service.

6.7.2 Auto Callback

The Auto Callback section of [RST TR] applies to BSS, with the following additions.

Multiple UEs with the same public user identity may be registered. In this case, if one of the UEs executes AC, then that UE uses the calling address received at that UE if the last received call was not anonymous. If the last call received was anonymous, the AR target address is the address (network stored) of the caller associated with the last received call at this UE.

6.7.2.1 Feature Interactions

The feature interactions of Automatic Callback feature are specified in [GR 215].

Calling Name Delivery

When AC is used with the CNAM feature, the called station's name need not be delivered to the customer when special ringing is applied. This differs from the requirements of [GR 215].

Call Forwarding

An AC request to a target with Call Forwarding (see Section 6.4) active should be rejected instead of being forwarded. The call to the line which originated the AC request (with the special ringing pattern) rings the line and is not forwarded.

Call Rejection

An AC request to a target with a service which rejects calls (e.g., Anonymous Call Rejection, Do Not Disturb) is rejected if a normal call from the subscriber to the target would be rejected. The call to the line which originated the AC request (with the special ringing pattern) is not rejected by any Call Rejection service.

Auto Recall

AC can be used to establish a call to the last AR target. Concurrent AR and AC requests from a user to the same target may be treated as one request.

Simultaneous/Sequential Ring

If the AC target is a Simultaneous or Sequential Ring subscriber, busy/idle monitoring only occurs on the subscriber's own line and not any of the other Simultaneous/Sequential ring destinations.

Hunt Groups

Multiple activations from different lines in a Hunt Group to the same target are allowed and are treated as separate AC requests. The call to the line which originated the AC request (with the special ringing pattern) does not alert other lines in the hunt group. It is an implementation choice whether AC requests which target a Hunt Group are accepted or rejected.

Call Pickup

The call to the line which originated the AC request (with the special ringing pattern) cannot be picked up from another line using Call Pickup. The call to the AC target can be picked up.

Outbound Call Blocking/Call Restriction

The call type assigned to AC calls will typically have the same classification as a normal call to the target and so may be denied by the Outbound Call Blocking/Restriction service.

6.7.3 Last Number Redial

The Last Number Redial feature enables a user to automatically redial the last called number by pressing a redial button on their telephone set. Alternatively, this feature could be invoked by dialing a Feature Activation Code.

Last Number Redial is distinguished from the Auto Callback feature described in Section 6.7.2 in that if the target public user identity is busy, Auto Callback performs camp-on processing, while Last Number Redial simply returns busy treatment to the originating user. As such, BSS does not support the option to activate Last Number Redial via feature activation code, since the Auto Callback feature provides the same functionality plus more. The client-based version of Last Number Redial is supported, although its operation is trivial and is not specified in PacketCable documents.

6.7.3.1 Feature Interactions

Network-based features are unable to distinguish between a user invoking the Last Number Redial feature and the user manually redialing the last number.

6.8 Miscellaneous Features

6.8.1 Do Not Disturb Client (DND Client)

The Do Not Disturb Client (DND Client) section of [RST TR] applies to BSS, with the following additions. It should also be noted that DND Client is within scope of BSS, even if the equivalent RST feature is noted as out of scope in [RST TR].

It is optional to allow the caller to announce their name to the called party who has the DND activated, thus providing the called party with the choice of accepting or rejecting the call.

6.8.1.1 Feature Interactions

Since DND Client is a terminating feature that is executed on the UE, all AS-based features will take precedence over DND Client.

6.8.2 Do Not Disturb Public User Identity (DND Public User Identity)

The Do Not Disturb Public User Identity (DND Public User Identity) section of [RST TR] applies to BSS, with the following additions/modifications.

DND Public User Identity does not put restrictions on the number of UEs that may be beyond a given public user identity.

It is optional to allow the caller to announce their name to the called party who has the DND activated, thus providing the called party with the choice of accepting or rejecting the call.

6.8.2.1 Feature Interactions

The feature has the following interactions.

Basic Calling Capabilities

UEs with DND Public Identity activated are able to originate calls.

Multi-Party Features

Blind Transfer recall results in alerting, even if the DND is activated.

Call History Features

Call-back features, such as Auto Callback and Auto Recall, alert the UE that the requested the Auto Callback and Auto Recall with DND is activated.

Call Forwarding

DND Public Identity takes precedence over Call Forwarding Variable.

DND Public Identity takes precedence over Call Forwarding to Voice Mail.

DND Public Identity takes precedence over Call Forwarding Don't Answer.

DND Public Identity takes precedence over Call Forwarding Busy.

Selective Call Forwarding takes precedence over DND Public Identity.

Selective Call Acceptance takes precedence over DND Public Identity.

Do Not Disturb

Busy Line Verify takes precedence over DND Public Identity.

Anonymous Call Rejection

DND Public Identity takes precedence over Anonymous Call Rejection.

DND Client

DND Public Identity takes precedence over DND Client.

6.8.3 Screening List Editing (SLE)

The Screening List Editing (SLE) section of [RST TR] applies to BSS.

6.8.4 Subscriber Programmable PIN (SPP)

The Subscriber Programmable PIN (SPP) section of [RST TR] applies to BSS.

6.8.4.1 Feature Interactions

Once SPP programming is started, flash processing by the UE is disabled only if SPP programming is performed via an analog telephony interface.

6.8.5 Distinctive Alerting

The Distinctive Alerting section of [RST TR] applies to BSS.

6.8.5.1 Feature Interactions

Call Waiting

When the Call Waiting feature is activated for the called UE and the BSS AS has access to the Call Waiting activation status, the BSS AS may select a special alert that can either provide distinctive ringing or a distinctive call waiting alert to the called party.

Caller ID Display Blocking

When the calling party has requested that his or her Caller ID not be displayed, if the call is determined by the distinctive alerting algorithm to receive a distinctive alert, and when the BSS AS is configured to give Distinctive Alerting precedence over Caller ID Display Blocking, then the BSS AS will provide the Distinctive Alerting Treatment.

When the calling party has requested that his or her Caller ID not be displayed, if the call is determined by the distinctive alerting algorithm to receive a distinctive alert, and when the BSS AS is configured to give Caller ID Display Blocking precedence over Distinctive Alerting, then the BSS AS will not provide the Distinctive Alerting Treatment.

6.8.6 Message Waiting Indicators

The Message Waiting Indicators section of [RST TR] applies to BSS, with the following additions.

The Message Waiting Indicator is restored upon any UE initialization (e.g., restart or reboot).

6.8.6.1 Feature Interactions

Do-Not-Disturb (DND)

If both DND and MWI are in effect, any audible alert for waiting messages should not be presented by the UE until the DND is deactivated, but a visual MWI can still be presented by the UE.

6.8.7 Speed Dialing

The Speed Dialing section of [RST TR] applies to BSS.

6.8.7.1 Feature Interactions

Outbound Call Blocking

If the originating UE subscribes to speed dialing and OCB, speed dialing needs to be invoked first so that the OCB service knows the actual destination number to check for blocking.

Three-Way Calling

Speed Dialing is usable for entering the called party ID during Three-Way Calling.

Call Forwarding

Speed Dialing is usable for entering the called party ID during various Call Forwarding features.

6.8.8 Customer-Originated Call Trace

The Customer-Originated Trace section of [RST TR] applies to BSS.

6.8.8.1 Feature Interactions

Auto Recall

If a new call is received but not answered, the embedded UE should not over-write the last answered calling party information, but rather should store this as a new un-answered call attempt. When the user invokes the AR feature, the UE uses the stored information (un-answered or answered call) with the most recent time stamp.

6.9 Fax, Voice-Band Data, and Other Media Types

6.9.1 Fax, Modem, and TDD Calls

The Fax, Voice-Band Data, and Other Media Types sections in [RST TR] applies to BSS, with the following additions/clarifications.

Pass-through mode is simply providing a G.711 (or another appropriate codec) media stream and allowing the fax, modem or TDD signaling to traverse that media stream without any additional transport robustness mechanisms (other than sending redundant G.711 media packets).

6.9.2 Alarm System Support

The Home Alarm System Support section of [RST TR] applies to BSS, with the following modification.

To avoid false alarms at the alarm system, it is also recommended that UEs embedded in cable modems with analog interfaces that connect to alarm systems follow the alarm-system-support requirements defined by PacketCable.

6.9.3 DTMF Relay

The DTMF Relay section of [RST TR] applies to BSS, with the following modification.

For BSS, we refer to [RFC 4733], which obsoletes RFC 2833 referred to in [RST TR].

6.9.4 DTMF in Audio

The DTMF Audio section of [RST TR] applies to BSS, with the following modification.

For BSS, we refer to [RFC 4733], which obsoletes RFC 2833 referred to in [RST TR].

6.10 Early Media

The Early Media section of [RST TR] and subsections apply to BSS.

6.10.1 Feature Interactions

Basic Call

The UE renders early media when early media is received, in lieu of local ringback.

DND PIN Override

DND PIN override uses two-way early media as described in Section 6.8.2 for announcement, and collection of PIN. The BSS AS supporting DND makes use of the procedures within [RFC 3959] in order to handle UEs that indicate support for the same.

Outbound Call Blocking

Outbound Call Blocking uses two-way early media as described in Section 6.5.1 for collection of overriding PIN. The BSS AS supporting OCB makes use of the procedures within [RFC 3959] in order to handle UEs that indicate support for the same.

6.11 Operator Services

The Operator Services section of [RST TR] applies to BSS.

6.11.1 Busy Line Verification

The Busy Line Verify section of [RST TR] applies to BSS, with the following clarification:

The BLV operator cannot monitor a busy call associated with the target user if the user is still able to accept new calls (e.g., busy user has call-waiting, or user is busy in a call on one UE but is able to accept new calls on another UE). This replaces the [RST TR] paragraph discussing the conditions when there is more than one UE behind a public identity.

6.11.1.1 Feature Interactions

Switch-hook flash is disabled in the target UE while the Busy Line Verification procedure is active.

Call Forwarding Variable

If the target UE has enabled Call Forward Variable to forward all calls, the Call Forwarding AS returns 486 Busy in response to the INVITE with JOIN header from the operator services UE. If the operator services UE receives 486 Busy as a response to an INVITE, the operator services UE plays the busy tone to the operator.

Do Not Disturb

If the target UE has enabled DND, the DND AS forwards the operator interrupt and BLV attempts to the UE.

Shared Call Appearance (SCA)

Shared call appearance (SCA) requires special handling from the BSS AS. When multiple UEs are registered on an identity, the BSS AS should deliver subscriptions for the dialog event package from the "Operator" identity to the GRUU of the UE active in the call. The BSS AS may, as an alternative, locally cache UE dialog event notifications from the UEs and send a locally generated NOTIFY to the "Operator" entity subscribing to the dialog event package. An INVITE with a join header from the "Operator" identity should be treated similarly. It should be sent to the GRUU of the UE active in the call. Handling for UEs bridged on a line is unspecified and, as an implementers option, requests that require more than three party mixing or conferencing may be allowed to fail.

6.11.2 Operator Interrupt

The Operator Interrupt section applies to BSS.

6.11.2.1 Feature Interactions

Feature interactions for Operator Interrupt are identical to those for Busy Line Verification defined in Section 6.11.1.1.

6.12 Emergency Services

The Emergency Services section and subsections in [RST TR] apply to BSS.

6.12.1 Feature Interactions

Since no AS are involved for emergency calls originated by a UE, there are no feature interactions for AS based provided services.

Terminating non-Emergency Call

When the UE is in an emergency call, the UE rejects all incoming calls that are not emergency calls.

Call Hold

The UE should not be able to initiate call hold while active on an emergency call.

Caller ID Per-Line Blocking

The UE should not block Caller ID when an emergency call is made. Although the Caller ID is not currently used by the PSAP, it may be used in the future.

Caller ID Per-Call Blocking

The UE does not block Caller ID when an emergency call is made. Although the Caller ID is not currently used by the PSAP, it may be used in the future.

Auto Recall

The UE and the Auto Recall AS do not allow the Auto Recall feature to be used to target an emergency phone number or address. The UE and Auto Recall AS should block Auto Recall of an emergency call with long term denial treatment.

Auto Callback

The UE and the Auto Callback AS do not allow the Auto Callback feature to be used to target an emergency phone number or address. The UE and Auto Callback AS should block Auto Callback of an emergency call with long term denial treatment.

Customer-Originated Call Trace

The UE and the BSS AS do not allow an emergency ringback or callback to be traced.

6.13 Monitoring and Management

Monitoring and Management section and subsections in [RST TR] apply to BSS.

6.14 Lifeline Capabilities

The Lifeline Capabilities section of [RST TR] applies to BSS.

6.15 Business Oriented Individual Line Features

This section describes individual or personal line (non-group) features that may need to be supported in order for cable operators to deliver business voice services to SMBs. These features are not typically supported in a residential environment.

6.15.1 Automated Attendant (Auto-attendant)

The auto-attendant function allows incoming calls to be automatically transferred to the appropriate extension or destination (e.g., voice-mail) without intervention of a human receptionist. The auto-attendant function typically utilizes an Interactive Voice Response (IVR) system to provide prompts to the originator, and then routes calls based on the responses provided to the prompts. The enterprise must have the ability to configure the prompts so that they can be customized to reflect individual business needs.

Calls typically route to auto-attendants in one of two ways. With the first approach, a "main number" is published for the enterprise which causes the network to route calls directly to the auto-attendant. The auto-attendant may lead the caller through a series of prompts and responses in an attempt to address the caller's need without involving a human receptionist in the conversation. If human intervention is ultimately required, though, the auto-attendant can transfer the call to another extension.

With the second approach, the caller may initiate the session by directly dialing a line within the enterprise, and if that line is not answered, the call is forwarded to the auto-attendant. This is especially common in scenarios where the auto-attendant also hosts voice-mail capabilities, and has behavior that aligns with typical voicemail scenarios. The auto-attendant may directly connect the user to a voice mailbox, or may provide additional prompts and information. Like the first approach, the auto-attendant can transfer the call to another extension if human intervention is required.

Support for automated attendant will focus on carrier-hosted solutions, where the attendant is deployed in the carrier's network, as opposed to being a CPE device.

Calls terminate to the auto-attendant via standard SIP signaling, and the auto-attendant answers the call. After the call has been answered, the auto-attendant may transfer the call to another destination using standard SIP signaling (such as the REFER method). Multiple simultaneous sessions must be allowed to terminate to automated attendant, with each session being handled independent of the others.

Use of auto-attendants is discouraged in configurations where parallel forking is leveraged. Auto-attendants typically answer sessions automatically when they are presented to them, making it virtually impossible for other forked sessions to have an opportunity to accept the calls. This can lead to undesirable side effects, such as a burst of ringing on a forked client when an INVITE is received followed soon thereafter by a CANCEL.

The auto-attendant use case is a common one that's used as an example in many IETF SIP drafts, although in the majority of cases the interconnect can be addressed in standard SIP through a combination of [RFC 3261] SIP and REFER.

The reference for this features is [GR 524] - Attendant and Customer Switching System Features and Customer Interfaces. It contains discussion on call setup, hunt group interactions, feature interactions, and billing requirements.

NOTE: [GR 524] also discusses requirements around DID Calling, hotel/motel calls, [GR 2917] discusses an ISDN Smart Attendant service. It discusses several attendant functions/services, but from the perspective of a "human" attendant, as opposed to an "automated" attendant. These include recall, night-service, busy-lamp notifications, serial calling, and camp-on.

6.15.2 Direct Inward Dialing, Direct Outward Dialing (DID/DOD)

All user terminals assigned an E.164 compliant directory number (DN) can directly send or receive calls to/from the PSTN unless restricted by call blocking features. Specifically, DID can be allowed or blocked on a user basis for the listed DNs associated with either Selective Call Acceptance (Section 6.5.3), or Selective Call Rejection (Section 6.5.4). Inward collect calls can also be blocked. DOD can be blocked on a user basis by invocation of the Outbound Call Blocking feature (Section 6.5.1). It will be possible to enable or disable DID and DOD on an individual DN basis.

6.15.2.1 Feature Interactions

DID is allowed or blocked on a user basis for the listed DNs associated with either Selective Call Acceptance (White List) or Selective Call Blocking (Black List). Inward collect calls can also be blocked.

DOD can be blocked on a user basis by invocation of the Outbound Call Blocking feature.

6.15.3 Extension Dialing (Private Numbering Plan)

A Private Numbering Plan (PNP) will be used to provide extension based and abbreviated dialing, allowing a Business Group user to call another user within the Business Group, or a predefined external destination. A PNP describes the dialing plan for a private telephone network. A PNP may offer other features, such as requiring one or more prefix digits to disambiguate between a number inside the PNP and a number outside the PNP (e.g., prefixing an external number with a "9").The PNP will map to the global E.164 Numbering Plan as partial numbers, as defined in [ITU-T E.164] Annex B. If DID/DOD is enabled for an extension, the partial number will be mapped to a full E.164 format. For extensions that do not have DID/DOD enabled, this is not required. Each Business Group can have a unique PNP.

All PNPs will use a fixed length of numbers. This supports fixed length E.164 numbers and a predefined digit map for terminals. The PNP can be either migrated from an existing numbering plan, in case of a PBX or Centrex migration or defined from scratch. In the case of PBX or Centrex it is assumed that the numbering plan is already compatible with E.164 and proper access codes to the local or national E.164 numbering plan exists. When new PNPs are defined, the following is recommended:

- Define the number of digits N of the local number. The minimum number of digits is 2. The maximum will be dependent on expected final number of users and the type of E.164 number to be used to address the BSS subscriber from E.164 (4, 5 or even 6 digits).
- No private number will start with digit 0.
- No number related to a user starts with 1 and 9;
- therefore private numbers should start from 2000 to 8999.
- The E.164 number requested to address the BSS as a whole will be according to [ITU-T E.164] Annex B.3.2 in such a way that the CC N(S)N SN or CC GSN including the private number as partial number gives a full E.164 number and does not exceed 15 digits (in national numbering plans additional restrictions may apply).

PNPs may provide network specific numbers for internal use only. These numbers will not be reachable from the outside, since they are not E.164 numbers.

An operator may decide on the use network specific numbers at his own discretion, but the following principles are recommended to be common for user convenience:

- Network specific numbers may be used for providing internal services specific for this BSS and providing access to emergency services.
- Network specific service numbers will start with digit 1 (or 9).

A web portal may provide a business group administrator with the ability to manage the PNP. This might include assigning "extension" codes to business group members, as well as the mapping between a short code and its destination (e.g., E.164 number).

6.15.3.1 Feature Interactions

Caller ID Delivery

When a call is presented to a user, the Calling Number Delivery service may substitute the caller's public directory number with a number from the Private Number Plan (e.g., so that the caller's extension is displayed).

Call Transfer

A user can transfer a call by entering a number within the Private Number Plan.

Outbound Call Blocking

Calls dialed using an extension will typically be subject to different Outbound Call Blocking rules than a fully dialed destination. For example, dialing another extension is typically permitted even if the call might normally be classified as a long-distance call and long distance calls are blocked.

Distinctive Alerting

When a call is presented to a user, the Distinctive Alerting service may provide a customized alerting behavior when the caller is another extension within the Private Number Plan.

Account Codes

Calls dialed using an extension will typically be classified as a different call type than a fully dialed destination. For example, a dialing another extension may not prompt for an Account Code even if the call might normally be classified as a long-distance call requiring an Account Code.

Call Pickup

A user can invoke directed call pickup by providing the extension of the ringing phone (rather than the full directory number).

Outbound Call Restriction

Calls dialed using an extension will typically be subject to different Outbound Call Restriction rules than a fully dialed destination. For example, dialing another extension is typically permitted even if the call might normally be classified as a long-distance call and long distance calls are restricted.

Emergency Calling

Depending upon the Private Number Plan's dial plan, the dialed digits used to make an emergency call may differ between subscribers. For example, if a user is required to dial "9" as an external line prefix, the user may need to dial "9911" to reach emergency services.

6.15.4 Alternate Number (Virtual Number, Second Number)

The Alternate Number feature allows users to have multiple directory numbers which all terminate on the same subscriber line. The main number assigned to the subscriber is referred to as the primary number. Implementations are expected to support a primary number and up to 9 alternate numbers per subscriber line. Implementations are expected to anticipate support for at least 20% of their dial plan being alternate numbers.

The primary number is expected to be a PSTN-routable E.164 number.

An alternate number can be any E.164 number. An alternate number may be in the same exchange <NPA-NXX>. An alternate number may be in a different exchange within the same area code. An alternate number may be in a different area code. An alternate number may be an international number that is not 10 digits. An alternate number may be part of a private numbering plan.

In any deployment where the alternate numbers are in a foreign exchange, there will be a donor network and a recipient network. The donor network is the switching environment that actually controls that block of telephone numbers. The recipient network is the switching environment that controls the client device that receives calls for that alternate number.

- The donor network might be controlled by the same service provider.
- The donor network might be another PacketCable service provider.
- The donor network might be some non-cable service provider with a business relationship with the cable operator.

Outgoing calls from the subscriber line are expected to use the Caller ID of the primary number assigned to the subscriber line.

6.15.4.1 Feature Interactions

A description of the most significant feature interactions is given here.

Audible notification (distinctive ringing)

- Each alternate number may have a unique distinctive ringing cadence.
- Alternate numbers for a subscriber line may be grouped to share a common ringing cadence.
- Business feature phones may provide custom ring tones or audio clips to identify each alternate number.
- Distinctive ringing cadences and custom ring tones are expected to be configurable through an administrative interface. They may also be configurable through a web portal.

Visual notification (called party number display)

- Business feature phones may provide a visual display indicating the called party number and text further identifying the alternate number.
- Business feature phones may provide unique lamp cadences when ringing for alternate numbers.

Call Forwarding, Sequential Ring, Find-Me/Follow-Me

- The enhanced capabilities of Alternate Number are not required to be preserved after call forwarding other than forwarding to voice mail as described below.

Voice Mail, Voice Response, and Unified Messaging

- The voice mail integration is expected to provide capability for a unique greeting per alternate number.

- Voice mail and voice response integrations are expected to provide a capability for separate languages per alternate number. For example, the primary number may use English prompts and an alternate number may use Spanish prompts.
- The voice mail integration is expected to provide capability to store messages to each alternate number in a separately accessible mailbox.
- When messages for primary and alternate numbers are all placed in the same voice mailbox, the voice mail system is expected to save the called number (primary or alternate) for each voice message. The user is expected to be able to access this information for each message.
- A unified messaging integration is expected to support the capability to route messages left at alternate numbers to different email addresses.

Key System Emulation / Shared Call Appearance

- If the stations are legacy analog telephones, the distinctive ringing cadences for alternate numbers are expected to function on all the stations.
- The default audible and visual notification for a line appearance on a feature phone may take precedence over audible and visual notification for the alternate number feature. For example, a line appearance on a feature phone may be configured to alert with a delayed ring burst if that line rings for 10 seconds. This delayed ring burst takes precedence over any distinctive ringing for alternate numbers.

Hunting

- If dialing the primary number results in the call being directed to a hunt group, any alternate number will receive the same treatment.

Call Pickup

- A subscriber line that is part of a pickup group will behave the same for both the primary number and any alternate numbers. With directed call pickup, an implementation may allow alternate numbers to be dialed to pick up the call. This is particularly useful with private numbering plans.

Call Park

- Similar to call pickup, an implementation may allow retrieval of a parked call to use alternate numbers. This is particularly useful with private numbering plans.

Billing

- The alternate number is expected to appear in the billing record as a supplemental field.

6.15.5 Intercom (Group Intercom, Push-to-talk)

The Intercom feature allows a user to place a call to a feature phone configured to auto-answer and enable the speakerphone for that class of call. Intercom calls may be placed by an analog telephone by dialing a private numbering plan number or by dialing a feature access code followed by a private numbering plan number. The private number may be the extension of a station or it may be a single digit that maps to a specific intercom group member. Only feature phones equipped with speakerphone can accept intercom calls.

The intercom feature may initiate a one-way talk path in the direction from the calling party to the called party. The called party then establishes a two-way talk path via a feature button on their telephone. The intercom feature may also be configured to auto-answer and create a two-way talk path. Feature phones can be configured to initiate intercom calls to another station via a single feature button. The functionality provided by the feature phone is vendor-specific but all cases need to be supported by the infrastructure.

NOTE: The Intercom feature does not support a point to multipoint configuration. The feature does not need to support delayed content since it is a real-time application. Thus, it does not require the technically complex Push-to-Talk solution.

Users and administrators can define accept and reject lists, which can contain wildcards. Administration of accept and reject lists can be done through the user and administrator web portals.

6.15.5.1 Feature Interactions

Do Not Disturb disables the intercom feature.

The intercom feature will take precedence over all other configured terminating call features such as:

- Call forwarding (and variants).
- Shared call appearance - only the primary will receive intercom calls.
- Sequential ring.
- Simultaneous ring.
- Hunt groups - pilot numbers are not allowable in the Intercom Group.
- Call waiting - no audible or visual notification will be given when a line is busy.

6.15.6 Shared Call Appearance (Multiple Appearance Directory Number - MADN)

This feature allows for a call appearance associated with a single address or directory number to be shared across multiple terminals. When an incoming call is received for this number, all the group members sharing the call appearance will simultaneously alert, and the first phone that answers the call is connected to the incoming call. This contrasts to other services such as simultaneous ringing in that this feature shares a DN call appearance between multiple terminals in the group, as opposed to simultaneously alerting a group of phones which each have their own directory number.

A common application of this service is to allow an administrative assistant to monitor the phones for a number of employees. Each employee's phone shares a call appearance with the administrative assistant's phone, which allows the administrative assistant to answer calls on behalf of each employee. Another potential application for this feature is implementing a hosted key system such that multiple lines are shared across multiple phones in an office.

Once a call is active with one group member, the other members of the group are not allowed to initiate or receive new calls on the shared appearance. The shared call appearance can be configured with an option to initiate bridging between the multiple group members (described in Section 6.15.7). If bridging is allowed, any member which shares the call appearance may bridge into the answered call, resulting in a conference between all parties involved in the call. If a terminal is not allowed to bridge into the existing call (as described in Section 6.15.7), it will receive NACK treatment from the network.

Inbound calls which terminate to a shared call appearance function in the following manner:

- The call originator initiates a session to the address (identity) of the shared call appearance.
- If there are not already active sessions on the shared call appearance, the call is offered to every terminal which shares the call appearance. Each terminal begins to alert.
- The terminal that answers first is awarded the call. The sessions which are being offered to the other terminals are canceled.
- The terminals which are not involved in the session may receive notifications to reflect the state of the shared call appearance. If bridging is allowed, the other terminals may attempt to bridge into the call at this point.

Outbound calls which originate from a shared call appearance function in the following manner:

- A terminal originates a session on the shared call appearance. The calling party information will reflect the identity of the shared call appearance.
- Only one terminal is allowed to initiate a call on the shared call appearance at a time. If another terminal attempts an origination while a call is already in progress, then the subsequent session will be rejected.
- When the call is answered, the other terminals sharing the call appearance may be allowed to bridge into the session.

Certain terminals can also support hold/retrieve function in combination with the shared call appearance. In those cases, a call to the shared call appearance DN can be answered on one terminal, held on that terminal, and retrieved from a different terminal in the group. Although this is similar to the capabilities provided by the call park service, there is a subtle difference in that the hold/retrieve action is performed between terminals that share a call appearance.

Whenever a call is offered or accepted on the shared call appearance DN, each of the terminals sharing the call appearance can be notified of the state of the call. Through these notifications, each terminal sharing the appearance can determine the status of the call associated with the appearance, and determine their ability to participate in the existing session or initiate new sessions. Depending on the capabilities of the terminal, this could take the form of lamps or some other type of display update. The following states should be displayed:

- Idle: there is no call on this appearance.
- Initiating: an originating call on this appearance is waiting for the remote user to answer. The call appearance transitions from "idle" to "initiating" when a user originates on an idle call appearance (say, user goes off-hook), and remains in this state until the call is answered or abandoned.
- Alerting: a terminating call on this appearance is waiting for the local user to answer. The call appearance transitions from "idle" to "alerting" when it receives a terminating call request, and remains in this state until the call is answered or abandoned.
- Active: there is an active call on this appearance (i.e., a call that was previously in the "initiating" or "alerting" state has been answered).
- Held: a previously "active" call has been put on consultative hold.

IETF work is underway in the BLISS Working Group to refine the behavior of this service.

6.15.6.1 Feature Interactions

Call Waiting

If one of the phones is already hosting an active call on the shared appearance, additional incoming calls are delivered to the active phone as a call waiting call. Only the active phone has the option of accepting or rejecting the call waited call. Generally accepting the call waited call precludes bridging into either call by other terminals which share the call appearance.

Three-Way Calling

If one of the phones is already hosting an active call on the shared appearance, that user can leverage the three-way calling service to initiate a new session. Invoking the three-way calling service on the active phone precludes bridging into either of the calls by other terminals which share the call appearance.

Call Forwarding

Call forwarding services (such as call forwarding unconditional, call forwarding busy, and call forwarding don't answer) are associated with the call appearance, as opposed to the individual terminals which share the call appearance. When these features are active, an incoming call destined to the shared call appearance will be redirected to a remote destination. When the call is redirected, it is no longer offered to any of the terminals which share the call appearance. When call forward unconditional is activated, the redirection takes place immediately on call terminations, and the call is not offered at all to the shared call appearance. When call forward busy is activated, the redirection takes place immediately if the call cannot be offered to the shared call appearance due to a busy condition. When call forward don't answer is activated, the redirection takes place after the shared call appearance has been alerted for a period of time, but an answer is not received from any of the alerting terminals.

Do Not Disturb (DND)

Whenever one of the terminals which shares a call appearance activates the DND feature, it results in that terminal to no longer be offered sessions which terminate to the appearance. Incoming calls will continue to terminate to the other terminals sharing the appearance. Once the call has been answered, the terminal with DND still receives state notifications associated with the active call, and is eligible to bridge into the session.

6.15.7 Shared Call Appearance Bridging (Multi-Location Shared-Line Appearance)

This feature is a variant on the standard Shared Call Appearance feature. Shared Call Appearance Bridging allows a second terminal to barge in ("bridge") onto an existing active shared call appearance.

If configuration allows shared calls to be bridged then a terminal that currently has a shared call appearance which has been notified of an active call can bridge onto that active call; thus establishing a multi-party call between the bridging terminal and the original call parties.

Each terminal that shares a call appearance has a provisioned data item that controls whether other terminals sharing the call appearance can bridge into calls to/from this terminal. This data item is provisioned to a value of "private" or "public", where bridging is blocked on private calls and allowed on public calls. The user of the terminal can override the provisioned value (flip it from public to private, say) on a per-call basis. Note that this data item must be maintained per public user identity per terminal (and not per public user identity across all sharing terminals).

Hosting of a bridged (and so multi-party conference) call appearance may be achieved by network located resources such as a conference server. This then allows multiple bridging attempts to be made on single shared call appearance up to a defined configurable limit. Should the bridging of a shared call appearance be limited to a single (1) bridged party then depending on the capabilities of the terminals that share the call appearance the bridging may take place within the device itself.

A shared call appearance can only be bridged when that call is stable; that is the call appearance state is active. Any other state for the shared call appearance result in the bridging attempt being rejected and an appropriate indication of such provided to the bridging terminal.

A BSS AS may optionally support client bridging, network bridging or both.

6.15.7.1 Feature Interactions

Multi Party Calls, Call Waiting

A shared call appearance can only be bridged when that call is itself not part of a multi-party call (multi-party calls include Call Waiting, 3WC and some instances of Call Transfer). For example, if a three-way call has been established then no bridging can take place, similarly if an active call also has a call waiting then no bridging can take place. In these circumstances, the BSS AS will respond to the bridging attempt with a 4xx response.

6.15.8 Sequential Ring

The Sequential Ringing feature enables a user (the callee), through call redirection, to receive calls at different locations. It allows a terminating subscriber to have the network redirect calls to a list of addresses (ringing list) which are called sequentially. Sequential Ringing is executed independently of the busy/idle state of the served subscriber. The served subscriber’s ability to originate calls is unaffected by activation of this service. A call is set up to each address from the ringing list one at a time for a defined number of rings. The capability to define this number per list entry is optional. If an entry in the ringing list is not reachable (e.g., a timeout or a busy situation occurs), that number is skipped and the service moves to the next list entry, e.g., the feature works through the rest of the configured ringing list.

When a user that is subscribed to the sequential ringing feature is called, the calling party may be informed about the feature progress via announcements. These progress-announcements are optional; for example, they might not be needed for the case where there are only a few entries in the ringing list.

Normally, this feature is assigned to a real public user identity associated with a real user. When a calling party somewhere in the global network initiates a call to this "real" public user identity, the sequential ringing feature initiates a series of sequential calls to each of the addresses on the sequential ringing list. The public user identity of the user that has subscribed to sequential ringing is the first entry on the ringing list (it is the Main DN). The list entries may point to a PacketCable user, or any E.164 address in the global PSTN or cellular network.

This feature may also be assigned to a virtual public user identity that simply points to a sequential ringing list containing a list of user addresses.

It is the responsibility of the feature user not to add entries to the ringing list that cause feature malfunction like loops, devices answering the call automatically or disturb other users and so on.

The sequential ringing feature provides a capability that enables the caller to interrupt the sequential ringing procedure at any time by entering a DTMF digit (e.g., to skip the remainder of the ringing list entries and route directly to voice mail).

The user has the capability to block some entries of the ringing list from feature execution, including the Main DN. It is left open to the user to add the address of his voice mail system as the last entry to the ringing list. The user shall be able to configure the ring duration per list entry.

Further options like rules to skip entries in the list of addresses based on time of day, date, and so on, are out of scope.

The feature by itself does not provide any criteria to select the calling user but can be combined with ACR or Solicitor Blocking for that purpose.

The Sequential Ringing feature is assigned to a PacketCable user and can be administered via a Web/Portal interface by the user, the business group owner or the operator. The feature activation/deactivation as well as the ringing list, blocking of entries etc. can also be administered via list editing through the telephone. This implies the use of respective service code procedures that may be executed from any DTMF capable phone.

The feature implementation and subscription shall reside in the BSS Application Server. SIP protocol issues are not foreseen except that early media situations might occur for calls to the user from PSTN.

There may be a list of numbers that are not allowed in the ringing list of any user - enforced by the network. For example, there are interactions to barring, closed user group settings and loop issues.

The sequential ringing procedure is stopped when the caller is connected to one of the numbers or all the numbers in the list are exhausted and no connection to any number on the ringing list could be established.

The feature can be applied to callers from any DTMF capable phones. The feature can be assigned to any BSS device - ideally with DTMF support.

The entries on the ringing list will be restricted to tel URIs (e.g., no alphanumerical addresses). The target addresses on the list may belong to the IMS or the CS/PSTN domain. In the latter case the diverted call is not forwarded to an IMS network but a terminating breakout is performed to the MGC.

The Sequential Ringing feature necessitates several announcements. The network will attach media server resources for the collection of DTMF key inputs.

[GR 568] defines a similar feature called Series Completion that allows calls to a busy Directory Number (DN) to be routed to another specified DN in the same switching office. This is a significant difference to Sequential Ringing as well as the fact, that the caller gets indications about the progress e.g., that hunting takes place and that the caller can influence the hunting during execution.

6.15.8.1 Feature Interactions

Note that some of the interaction cannot be enforced by the BSS-AS if the user enters DN's of the CS/PSTN domain to the ringing list.

Autoanswer systems (VMS, Auto Attendant, Load speaker paging and PTT)

Such devices / DNs shall not be entered to the ringing list.

Emergency

Emergency short cuts shall not be entered to the ringing list.

VSC - codes in general

Access and authorization codes shall not be entered to the ringing list.

Sequential Ringing

When processing a Sequential Ringing list, if a DN on the list has the Serial Ringing feature, that feature will not be started. However, the Sequential Ringing list from the owning DN will continue to be processed.

Teleworker / remote calls

Sequential Ringing is not applied to the A-party.

Do Not Disturb (DND)

Network DND takes precedence and the call may be re-directed acc. DND. If a list DN has DND active, then alerting for the Serial Ringing does not occur.

Call Forwarding

If the Main DN has a CF feature active, then the call will be forwarded based on the calling party and the Sequential Ringing feature will no longer be active for that call.

Call Waiting

If a target with Call Waiting does not respond within the Sequential Ringing time, then Sequential Ringing continues to the next target in the ringing list.

Executive Override / Busy Line Verification / Operator Interrupt

Sequential Ringing is not invoked on such calls, regardless of the busy or idle state of the called party.

Hunt Group

When a line with Sequential Ringing is reached via hunting, the Serial Ringing feature is not applied on the call.

Automatic Recall / Auto Call Back / Last Number Redial

If the called station has the Sequential Ringing feature, and the called party does not answer, the Sequential Ringing feature is used which will attempt terminations to the numbers on the list.

Call Park

The Sequential Ringing feature is not invoked, when the original parking user with the Sequential Ringing feature is rung back to remind them about a parked call.

Outgoing Call Restriction / OCB

When editing the Serial Ringing list the OCR/OCB lists shall be checked. During a call being originated by a subscriber with the two features, OCR/OCB lists are checked as normal.

Anonymous Call Rejection / Solicitor Blocking

If the Main DN or list entries have ACR active and the calling party number is anonymous, the Sequential Ringing feature is no longer active for that call respectively the list entry is skipped.

Selective Call Acceptance/ Selective Call Rejection

The Sequential Ringing feature is no longer active when the Main DN receives calls not allowed in the SCA / SCR lists or a PIN is not entered correctly.

6.15.9 Simultaneous Ring

The Simultaneous Ringing feature allows a user to be simultaneously rung at multiple locations. The Simultaneous Ringing feature is assigned to a PacketCable user in the form of a simultaneous "ringing list" containing a list of addresses to be alerted simultaneously when an incoming call arrives for the subscribing user.

Simultaneous Ringing is a terminating service that is optionally executed:

- if the state of the user is different from busy, or
- independently of the busy/idle state of the served device.

The list entries may point to a BSS user, or any E.164 address in the global PSTN or cellular network. Upon answer, the calling party is connected to the device that has answered the call first and the remaining ringing list entries are disconnected from the call. The served subscribers ability to originate calls is unaffected by activation of this service.

All phones ring until one is answered, or until timer expiration occurs (at which point another service such as Call Forwarding No Answer, Section 6.4.2, may be invoked). In contrast to Sequential Ringing (Section 6.15.8), the caller is not typically played an announcement.

Normally, this feature is assigned to a real public user identity associated with a real user. When a calling party somewhere in the global network initiates a call to this "real" public user identity, the feature initiates simultaneous calls to each of the addresses on the simultaneous ringing list.

A virtual number variant may be supported by associating the feature with a Public Service Identity.

It remains in the responsibility of the feature user not to add entries to the ringing list that may cause feature malfunction like loops, answering the call automatically, or disturb other users and so on.

If no connection to any member of the ringing list could be established, subsequent forwarding services may be used to forward the call to a mailbox service. The user may add an entry pointing to a voice mail system to the ringing list.

The feature can be administered via a Web/Portal interface by the user, the Business Group Administrator or the operator. The feature activation/deactivation as well as the ringing list can also be administered via screening list editing through the telephone. This implies the use of respective service code procedures that may be executed from any DTMF capable phone.

It should be noted that the feature can also be deployed to subscribers which are not part of a Business Group with slight variations interactions. This is out of the scope of this technical report.

The feature implementation and subscription shall reside in the BSS Application Server. SIP protocol issues are not foreseen. Several access codes are needed to activate/deactivate the service and edit the ringing list.

The network redirects calls to a ringing list of addresses which are all called in parallel. The first address that responds with a final successful response will be connected with the originating subscriber.

The service has to check feature interaction on the entries of the Ringing List with respect to barring, closed user groups settings, loop prevention, MLHG and so on. Cascading of Simultaneous Ringing is not supported. Based on information out of the History header subsequent parallel ringing services should be blocked.

The feature can be assigned to any BSS device - ideally with DTMF support to allow configuration over the telephone interface.

6.15.9.1 Feature Interactions

Note that some of the interaction cannot be enforced by the BSS AS if the user enters DNs of the CS/PSTN domain to the ringing list.

Autoanswer systems (VMS, Auto Attendant, Load speaker paging and PTT)

Such devices / DNs shall not be entered to the ringing list.

Emergency

Emergency short cuts shall not be entered to the ringing list.

VSC - codes in general

Access and authorization codes shall not be entered to the ringing list. Exclusion of features caused by this rule will not be mentioned in the following list.

Simultaneous Ringing

If a DN on the list has the Simultaneous Ringing feature, that feature will not be started.

Teleworker / remote calls

Simultaneous Ringing is not applied to the A-party of a 3PCC call.

Do Not Disturb (DND)

Network DND on the Main DN takes precedence and there is no ringing of the list DNs. If a list DN has DND active, then alerting of that DN for the Simultaneous Ringing does not occur, and no re-direction of the call to this DN to the DND intercept occurs.

Call Forwarding

CF takes precedence over Simultaneous Ringing.

Call Waiting

If the subscriber is busy with a call or another call is already waiting, then Simultaneous Ringing may not take place according the options mentioned above.

Executive Override / Busy Line Verification / Operator Interrupt

Simultaneous Ringing is not invoked on such calls, regardless of the busy or idle state of the called party.

Automatic Recall / Auto Call Back / Last Number Redial

If the called station has the Simultaneous Ringing feature, and the called party does not answer, the Simultaneous Ringing feature is used which will attempt terminations to the numbers on the list.

Hunt Group

When a line with Simultaneous Ringing is reached via hunting to the Pilot DN, the Simultaneous Ringing feature is not applied on the call.

Outgoing Call Restriction / OCB

When editing the Simultaneous Ringing list the entries shall be validated against the OCR feature data. During a call being originated by a subscriber with the two features, OCB is checked as normal.

Anonymous Call Rejection (ACR)

ACR takes precedence over Simultaneous Ringing. If one of the DNs on the Simultaneous Ringing list has ACR, that DN is not rung.

Selective Call Acceptance (SCA)

SCA takes precedence over Simultaneous Ringing.

Selective Call Rejection (SCR)

SCR takes precedence over Simultaneous Ringing.

Shared Call Appearance (SCA)

If the called DN has the Simultaneous Ringing feature and is an SCA DN, then all appearances of the SCA DN will ring. If one of the SCA DNs on the Simultaneous Ringing list is an SCA DN, then all appearances of the DN will ring.

6.15.10 Call Push, Call Pull (Call Jump)

Call Jump is a feature that enables the continuity of a call between defined devices associated with a subscriber. The call can either be "pulled", that is the initiation takes place from the 'move to' device or "pushed" that is the initiation takes place from the 'move from' device.

From a user perspective it allows procedures for a handover of calls between PacketCable devices and a standard cellular device, to which the user may be subscribed through any Mobile Network Operator (MNO). No pre-existing relationship is required between the MNO and the Cable MSO except that its network is reachable via the MGC. All roaming issues of the mobile device remain transparent to the Cable MSO.

A Call Jump subscriber is associated with a single 'Jump Group'; a subscriber cannot be part of more than one 'Jump Group'. This is a per subscriber set of devices between which a Call Jump can be invoked. The 'Jump Group' contains at least one cellular network device most commonly the subscriber's mobile with a subscription with a carrier other than the Cable MSO. However the 'Jump Group' could potentially have multiple wire-line devices with service provided by the Cable MSO.

Invocation of a Call Jump (either pull or push) is only possible from a device which receives service from the Cable MSO. The feature itself is invoked as follows:

Call Push: via a user trigger on a device on which a call is active and which has a valid Call Jump subscription the current active call is held. A new call leg is then extended to the target (push-to) device. The target device is identified either by pre-configured data, identified by a speed dial type short code or entered by the initiating user as part of a feature invocation sequence. On establishment of the new call leg the Call Jump is viewed as successful and the initiating device drops out of the call, leaving an established call and media path between the target device and the original call other party. Whether establishment of the extended call leg to provide Call Push is viewed as being on alerting or after answer should be determined via feature configuration or as a feature deployment option.

Call Pull: via a user trigger on a device which is currently idle and which has a valid Call Jump subscription invocation signaling is provided to the network. This signaling identifies that target device on which there is an active call to be 'pulled'; the signaling may also provide call identifying information such as SIP dialog if that is available to the initiating device. The network verifies the validity of the invoking device and whether it is permitted to 'pull' a call from the target device. Note that assertion of identity alone may not be sufficient especially if Call Pull can take place from any user associated device including those without Cable MSO subscriptions. The currently active call is extended to the initiating device and once established it drops out, leaving an established call and media path between the initiating device and the originally called other party.

A Call Jump attempt is viewed as having failed if for example, the new extended call leg in a Call Push is not established successfully, an entered target DN is not part of the user's defined Call Jump Group or if validity checks on a Call Pull attempt in the network fail. In initiation attempt failure cases appropriate indications of failure should be provided to the initiating subscriber.

The requirements to enable the application of Call Jump to calls involving non-Cable MSO provided cellular devices are worthy of special consideration.

To handle successfully and seamlessly the Call Jump of an active incoming call to the Call Jump subscriber the subscriber must also be subscribed to a Simultaneous Ring service where one of the Simultaneous Ring targets is the E.164 number of the cellular device that is part of the subscriber's Call Jump Group configuration. Thus a call to the subscriber that is answered on the cellular device will be anchored by the Cable MSO network and so subject to call leg manipulation as appropriate to achieve a Call Jump. Calls placed directly to the E.164 number of the cellular device that therefore are not handled by the Cable MSO network as part of call establishment are not subject to Call Jump. However such calls can be transferred from the cellular device via existing cellular call transfer mechanisms (such as ECT in GSM or XFR in CDMA) by the subscriber if required.

For successful and seamless handling of Call Jump of an active outgoing call initiated from the cellular device that is part of a subscriber's Call Jump Group, that call must be anchored within the Cable MSO network. This may be achieved via a number of mechanisms including providing a PSI (e.g., an 800 number) for call initiations from the cellular device which enables calls to be on-net routed, CAMEL or WIN triggers in the cellular network to route all originations via an MGC in the Cable MSO network or adoption of IMS Centralized Service (ICS) solutions (reference [TR 23.892], [TS 23.292]).

The feature execution is restricted to a single cellular device. The cellular device has no subscription within the Cable MSO network. It is known by the network only as an E.164 DN that is associated with the Call Jump user.

The subscription to Call Jump encompasses the capabilities for both Push and Pull.

The Call Jump feature in both its Pull and Push variant is restricted to a subgroup of devices in a Business Group defined by the operator or Business Group Administrator. This subgroup of devices is associated with the basic cable device the Call Jump user possesses, e.g., their default office phone.

The authorization to define the Call Jump list is not with the Call Jump user to avoid abuse - but with the Business Group Administrator.

The devices of this subgroup are numbered and can be addressed by the user dialing a short digit as an index instead of a full E.164 number or a PNP extension.

6.15.10.1 Feature Interactions

Simultaneous/Sequential Ring

The Call Jump subscriber is also required to be a Simultaneous/Sequential Ring subscriber such that all calls to their identity alert all devices within the Call Jump group.

Call Transfer

As Call Push is a specialization of Call Transfer then all of the feature interactions associated with Call Transfer are inherited by Call Push.

6.15.11 Remote Office (Satellite Office, Road Warrior)

Remote Office provides a user with the means to access all of the voice services and calling features that are available on his Business SIP Services line from a remote location that is equipped only with a computer that has Internet access, and a telephone that is capable of receiving incoming calls directly (which may be a cell phone). This feature does not require that the phone is provided by the SMB’s service provider or that the Internet connection has some minimum level of quality of service.

The Remote Office feature comprises two distinct capabilities:

- A call forwarding capability which ensures that incoming calls to the user's normal office phone are sent to the remote phone; and
- A third-party call control capability that provides Click-to-dial functions where the remote phone is temporarily treated as the user's phone for the purpose of originating calls. This capability may also offer other Third Party Call Control functions such as Call hold / retrieve.

To initiate and configure the Remote Office feature, the user enters the number of the remote phone into his computer. The Remote Office feature involves the establishment of a session with a Web browser or a local application, and when the session times out or the user logs out of the session explicitly; the Remote Office feature is automatically disabled.

The use of the Remote Office feature involves the establishment of an additional call leg between the system that hosts the Business SIP Services and the remote phone for each inbound and each outbound call that is made using the feature. A Call Detail Record is generated for each such call leg, so that it may be billed to the user.

6.15.11.1 Remote Office Call Forwarding

When the Remote Office feature is enabled, all incoming calls which would normally cause the user's phone to ring are diverted immediately to the remote phone number. When enabled, the user’s normal phone cannot be used to accept incoming calls to the user’s public user identities (phone numbers). This feature does not affect the ability for the user’s normal phone to accept incoming calls for a different public user identity assigned to the phone, for example using the Hoteling (Hot Desking) feature.

Many of the user’s terminating call control features operate normally when the Remote Office feature is enabled and can be controlled by the user using their personal computer (e.g., via the Web Portal). These features include:

- Call Forwarding - No Answer, Busy, Selective
- Selective Call Acceptance / Rejection
- Anonymous Call Rejection
- Do Not Disturb Public User Identity
- Simultaneous Ring
- Sequential Ring
- Incoming Call Notification and Control

The operation of other terminating features such as Calling Name/Number delivery and Distinctive Ringing is determined by the services offered at the remote phone number.

The call leg to the remote phone has diversion inhibited. This means that, if the remote phone is busy or doesn't answer, the call will be forwarded to the messaging system associated with the user's phone, not the remote phone. No ringsplash is applied to the user's phone when calls are forwarded to the remote phone.

6.15.11.2 Remote Office Click-to-dial

When the Remote office feature is enabled, an attempt to initiate a Click-to-dial call will cause the remote phone to ring so as to establish the first leg of the call. When the user answers the remote phone, the target destination number is called. The Caller ID that is presented to the called party is that of the user's own phone, not that of the remote phone.

The Caller ID that is presented on the first call leg (to the remote phone) is the user's own phone number, not the number of the target destination. This behavior is different from that which applies when Click-to-dial is used without the Remote Office feature. The reason for this is to discourage malicious use of Remote Office Click-to-dial to set up calls between two unsuspecting parties. By displaying the user's own number to both parties on the call, the identity of the initiator of the call is apparent. To ensure that a malicious user cannot conceal his identity by blocking delivery of his Caller ID, the user's own number is always sent as Caller ID to the remote phone, regardless of the status of his Caller ID blocking service. This cannot be construed as a breach of the service provider's obligations with regard to privacy, since by making use of the Remote Office feature, a user is implicitly indicating that he himself is the intended recipient of the first call leg to the remote phone.

6.15.11.3 Remote Office Third Party Call Control

Once a call has been established at the remote phone, whether it was an incoming call to the user's own phone that was forwarded, or a Click-to-dial call that was initiated by the user, any Third Party Call Control functions that are normally available to the user can be applied to the call. The user can therefore invoke Call Hold / Retrieve, Call Transfer, Conferencing and Call Release just as if the call was connected via the user's own phone.

6.15.11.4 Feature Interactions

The Remote Office logic is typically invoked after other terminating call features; the remote phone rings whenever the user's normal desk phone might ring. The feature interactions differ slightly from Call Forwarding Variable. For example, if the call isn't answered or the remote phone is busy, Call Forwarding Busy Line/No Answer is invoked. The Remote Office feature is unaffected by terminating call features implemented on the UE (e.g., Do-Not-Disturb Client, Call Waiting). The BSS AS does not allow the remote phone to be a blocked destination.

The remote phone may have its own terminating call services (e.g., call forwarding, call rejection, call waiting and distinctive ringing). These are invoked as normal and are unaffected by the fact that the Remote Office feature is being used.

6.16 Business Oriented Group Line Features

Group Line features are typically offered by a group phone system such as a Key Telephone System or PBX. Group features are designed to address the more advanced telecommunications needs of a group of users in an individual business or department who may need to share communications equipment and facilities in order to work collaboratively with colleagues via the telephone system. For example, consider a small auto dealership, which might have an accounting department, sales department, parts department, and service shop. Instead of dedicating an individual telephone line to each user, the dealership requires solutions that allow users to share lines. Additional features that support the unique needs of individual departments within the organization are needed, such as extension dialing and accounting codes. Hunt groups and other call distribution features that automatically distribute inbound calls to the appropriate person a department may be needed.

6.16.1 Account Codes

The Account Codes Service allows the subscriber to enter a special digit sequence (the account code) that is reflected in the billing record generated for the call. This enables the tracking and categorization of calls made or received by the subscriber. Account codes are managed by a Business Group Administrator and can be from 2 to 14 digits long. Account codes can be variable length, or configured to be a fixed length as a part of the configuration data for the service.

The subscriber may associate an account-code with a session using one of several methods. In the first method, the subscriber enters a vertical service code (*VSC) to initiate the account code service. The service then provides recall dial tone which prompts the subscriber to enter the account code. The end of the account code is indicated in one of three methods:

- by either having the subscriber enter a '#' digit
- the expiry of interdigital timing
- when sufficient account code digits have been entered, in scenarios where fixed length account codes are configured.

After the account code has been entered, dial tone is provided to the subscriber and they can proceed with dialing the called DN for the session.

In the second method, the subscriber can perform a mid-call activation of the service. When this is done with an analogue POTS set, the subscriber initiates the account code service by entering a switch-hook flash on an active call. After receiving recall dial tone, the subscriber dials the VSC associated with the service. The service provides recall dial tone, and the subscriber enters an account code. The end of the account code is determined the same as in the first method. After the account code has been entered, dial tone is provided to the subscriber, and they can take one of two options.

- If the intent is to associate an account code with the initial leg of the call, then the user may enter a switch-hook flash and be re-connected to the initial call. The account code entered will be included in the billing record for the call.
- If the intent is to have the account code apply to the second leg of a three-way call, then the user can proceed with dialing the DN of the party to be added to the conference. From this point forward, the call proceeds as a normal three-way call. The account code entered will be included in the billing leg for the second leg of the three-way call.

Mid-call activation can also be performed on a more sophisticated SIP business set in such a way that it does not interrupt the session that is in progress. The business set would provide an interface for the user to specify the account code and indicate which session should be associated with the account code. The application server can be notified of this with call control signaling, in such a way that the active session does not need to be placed on hold.

A third method for initiating the account code service is by configuring the user's profile or dialing plan to indicate that account codes are mandatory either on all calls, or on all calls to specific destinations. In these scenarios, the user begins by dialing the destination DN. After the application server determines that an account code is mandatory, dial tone is re-applied to the user's line which prompts them to enter the account code digits.

A web portal may provide a BG administrator with the ability to define account codes attributes for the BG (length, fixed or variable) and manage the user profile for the third method.

The service does not provide validation of the account codes, aside from insuring that account codes have a sufficient number of digits if fixed-length account codes are configured. If an invalid account code is detected, it is flagged in the billing record associated with the call.

The Telcordia reference for this service is [GR 605], although it provides no guidance in the context of mid-call activation of the service.

6.16.1.1 Feature Interactions

When used in combination with an analogue POTS set, Account Codes will have certain limitations when interacting with other services. For example, the account code service cannot be activated using the flash if other flash-driven features (such as call waiting or three-way calling) are active. In the case of call waiting, the subscriber is not allowed to use the flash method to enter an account code after the subscriber has already accepted the second incoming call. In the case of three-way calling, the subscriber is not allowed to use the flash method to enter an account code after the subscriber has already initiated the consult leg of the three-way call. Note, though, that an account code can be entered for the second leg if the VSC for the service is dialed prior to dialing the destination digits on the consult leg.

6.16.2 Authorization Codes

The authorization code service allows the subscriber to enter a special digit sequence (the authorization code), and thereby replace, for a single call, the restrictions assigned to the user with the restrictions assigned to the authorization code. Authorization codes can be from 2 to 14 digits in length, and are managed by a group administrator.

The authorization code service can be used to override Outbound Call Restrictions. Authorization codes can be unique to a given user or shared within a department, and different codes (each which facilitate different dialing privileges) could be dialed from any line.

The authorization code service can be triggered in one of two methods. In the first method, the subscriber enters a vertical service code (VSC) to initiate the authorization code service. The service then provides recall dial tone which prompts the subscriber to enter the authorization code. The end of the authorization code is indicated by either having the subscriber enter a '#' digit, or the expiry of interdigital timing. If the authorization code is valid, the subscriber is prompted with dial tone to enter the destination digits for the session. The destination digits will be translated and routed based on the restrictions that have been associated with the authorization code.

In the second method, the subscriber's profile indicates that an authorization code is mandatory. The subscriber begins by dialing the destination digits of the session. After the digits have been received, the subscriber is prompted with dial tone to enter an authorization code. The end of the authorization code is indicated by either having the subscriber enter a '#' digit, or the expiry of interdigital timing. If the authorization code is valid, the destination digits will be translated and routed based on the restrictions that have been associated with the authorization code.

For both methods if the subscriber does not enter a complete authorization code, or if the authorization code entered is not valid, reorder tone should be applied.

The authorization code service is somewhat similar to the subscriber activated call blocking feature, but there are key differences that can be used to distinguish them from each other. Subscriber activated call blocking leverages a PIN to override screening capabilities associated with the user's profile, and generally a single PIN is associated with each user's line and is not shared between users. Authorization codes can be unique to a given user or shared within a department, and different codes (each which facilitate different dialing privileges) could be dialed from any line.

The Telcordia reference for the authorization code service is [GR 605].

6.16.2.1 Feature Interactions

Authorization code overrides Outbound Call restrictions.

6.16.3 Hunting and Hunt Groups

Hunting is a terminating service that selects an idle (non-busy) member of a provisioned group of users (the hunt group) to receive an incoming call. Inbound calls to the pilot number of the hunt group are distributed across the members of the group using a specific hunting algorithm. Ultimately one idle member of the group is selected to receive the call, and the call is offered to that member. If a member of the group is busy, hunting proceeds to select the next available member to receive the call. In scenarios where all members of the group are busy, the group can be configured with an overflow destination or treatment to which the incoming call is routed. A user can be a member of multiple hunt groups.

The following hunting group algorithms should be supported.

- Circular - sends calls in a fixed order. The call is sent to the first available person on the list, beginning where the last call left off.

  This aligns to the service description in [GR 569].

- Sequential/Regular - sends calls to users in the order listed by an administrator. Incoming calls go to the first available person on the list, always starting with the first person on the list.

  This aligns to the service description in [GR 569].

- Directory Number Hunting (DNH) - A DNH group is comprised of users who each have a unique directory number, as opposed to other hunting algorithms where all calls come in using a single directory number of the hunt group. Whenever the directory number of a DNH member is dialed, hunting begins at that member, and proceeds sequentially through the group until an idle member is located. Hunting proceeds in a circular fashion until an idle member is identified.

Even though a SIP client may be capable of handling multiple calls at the same time, service requirements in [GR 569] stipulate that the hunting service should only offer a call to an idle member (one which is not currently involved in a call). It is possible that the UE can host additional identities (referred to in [GR 569] as non-hunt DN) independent of the hunting service. Active sessions associated with those identities are independent of the hunting service and do not factor into the BSS AS logic with respect to offering a hunting session to the UE.

The following hunting group algorithms may be supported.

- Weighted Call Distribution (Uniform) - enables calls to be distributed to agents according to a pre-defined weighting. Each agent is assigned a weight corresponding to the percentage of incoming calls they should receive.

  The "weighted call distribution" algorithm actually goes beyond the description in [GR 569]. The GR discusses "uniform call distribution", which could be considered a special case of weighted call distribution

- No Answer Policy - Business Group Administrators can also establish a No Answer Policy to redirect calls to the next agent if not answered in a specific number of rings by the previous agent. If all idle phones have been visited once without answer, there are two options for handling the call: forward call to an external number, or give the call a Temporarily Unavailable treatment, which can trigger a service such as voice mail.

- Selective Hunting Based on Calling Number - The ability to route calls to actual agents within a hunt group based on the calling number. For example, calls from a certain region, as indicated by the NPA-NXX, could be routed to distinct agent extension.

The implementation of hunt groups typically doesn’t include the ability to queue calls whenever all members of the group are busy. Queuing mechanisms are typically associated with more sophisticated services such as automatic call distribution, as opposed to general hunting capabilities. Hunt groups can be configured with overflow directory numbers or routes which will be used to route traffic in scenarios where all members of the hunt group are busy.

The Telcordia reference is [GR 569] Multi-Line Hunt Service.

The BSS description of "circular" and "sequential/regular" algorithms align well with the [GR 569].

The [GR 569] describes a capability called "preferential hunting", which effectively creates a subset of the overall hunt group that is searched first before the call is offered to the remaining members of the group. This is not in scope but can effectively be supported via the hunt group overflow mechanism.

Another common hunting algorithm is Directory Number Hunting (DNH). A DNH group is comprised of users who each have a unique directory number, as opposed to other hunting algorithms where all calls come in using the directory number of the hunt group. Whenever the directory number of a DNH member is dialed, hunting begins at that member, and proceeds sequentially through the group until an idle member is located. Another variation is where each member is assigned their own subset of other members to hunt.

Individual hunt group members would always have an address in order to register so that the core can route to them when selected. This address is unique and different from the address used to route calls into the hunt group. Whether this address is well known and callable directly or not callable directly is controlled via operator provisioning.

A client device that has the ability to support multiple DNs (e.g., multiple DN keys or lamps) should be able to display to the user whether an incoming call is addressed to the group or the individual member.

No specific IETF RFC or drafts applicable to the service have been located. The general consensus in the industry is that hunting is a service provided by the network server, and should not have any implications on the SIP protocol used to the client. In practice, variations call hunting algorithms are common capabilities provided by SIP soft-switches, IP-PBXs, and network call servers.

When the hunting service is extended to SIP clients, the service logic must account for the fact that certain client-based services can result in the client appearing to be busy, but not actually involved in an active session. In those scenarios, the hunting logic should be able to respond to this by proceeding to the next member in the group until an idle candidate is identified.

6.16.3.1 Feature Interactions

No Feature Interactions identified.

6.16.4 Call Park and Directed Call Park

The Call Park and Directed Call Park features enable a user to place a call on hold from one extension, and to subsequently retrieve it from another extension within the Business Group. The call is parked against a Park-Identifier which is used to retrieve the call. The park-identifier may correspond to a physical, or may be a virtual identifier.

- For Call Park, the user does not provide a park-identifier with the call-park request. The call is parked against a default park-identifier, often the user's extension.
- For Directed Call Park, the user provides a park-identifier with the call-park request.

The Call Retrieve function enables a user to be connected with a call that is parked.

Only one call can be parked at a time against a park-identifier.

Any extension within the group - including the original extension that parked the call - may retrieve a parked call.

A number of user interfaces are possible, each with their own usability trade-offs. [GR 2913] describes one analog interface variant for the Call Park features.

This section describes the case where Park-identifiers correspond to a physical extension as per [GR 2913].

An analog device user activates Call Park by dialing a vertical service code (VSC) according to the procedures described in Section 7.1. For Directed Call Park, the user follows the VSC with a park-identifier. For example, the user could go off hook and dial '*77' + <park-identifier> to initiate the Directed Call Park feature. For Call Retrieve, the user may follow the VSC with an optional park-identifier. Separate VSC values would be defined for Call Park, Directed Call Park and Call Retrieve. Alternative user interface may be provided for non analog devices. However the UE to network interface signaling used is the same.

For UEs that do not have an analog phone interface, the analog user interface requirements of [GR 2913] may be replaced with user interactions that provide similar results such as pressing a button on the device or clicking a virtual button on a screen. A visual reminder of calls that are parked may be provided. When an extension number must be specified, such a UE may allow the extension number to be entered either on a phone keypad or in a device-specific manner.

While a call is parked to an extension, calls can be made and received normally on that extension except that a new call cannot be parked against that same extension until the parked call is retrieved from the extension.

Only one call at a time can be parked against a park-identifier. If an attempt to park a second call is made then 2 seconds of reorder tone should be applied and then the original call between the parking party and the remote user on consultation hold should be re-established. If an attempt to park a second call to the parking user's extension is made by hanging-up after dialing the feature access code, the call will be re-offered immediately. If an attempt to park a second call is made by a UE with a non analog phone interface then an audio and/or visual indication of failure should be given and the original call should be re-established.

The parked call will hear music if the parking UE is a member of a group that is subscribed to the Music on Hold feature.

This section describes the case where Park Identifiers correspond to a virtual extension / general parking area, sometimes also called "park orbit".

The user enters a virtual identifier instead of an extension.

For the analog phone interface the system can optionally select a free park-identifier and play an announcement to indicate where the call ended up.

For the special case where a Business Group has only a single park orbit, black-phone operation is simplified. To park or retrieve a call, the user simply dials the park feature access code (without the need to also enter an extension number).

The user interface for non analog phones is not specified. As an example it is envisioned that a few physical keys (or virtual on-screen keys) may be assigned virtual extensions shared by all employees in the group; in this case one-touch park might operate as follows:

- One user parks the call on "park 1" by pressing the "park 1" key and then hangs up.
- The "park 1" key lights up on all phones - showing that there is currently a call parked there.
- Another user retrieves the call on a different phone by picking up the handset and pressing the "park 1" key.

The optional Timed Recall feature ensures that a parked call is re-established within a predefined time. If the call is not retrieved within this period of time, the call will be re-offered to the parking UE (not the extension it was parked to, if they are different). If the UE is busy, does not respond, or rejects the call, the call is returned to its previous parked state. The call will be reoffered for a maximum of three times. If the UE does not respond the third time, the call will be cleared or optionally forwarded to a specified number.

6.16.4.1 Feature Interactions

- A BSS AS may offer a call originated by Timed-recall to a user with SCA or SCR or DND-Public User Identity active.
- A BSS AS may apply Music On Hold to a parked call.
- The UE rejects a Timed-recall when DND-Client is active.
- The UE may apply Distinctive Alerting to a Timed-recall.
- The UE may apply Caller ID Delivery to a Timed-recall.

6.16.5 Call Pickup Features

[GR 590] describes the analog phone interface for the Call Pickup Features.

The Group Call Pickup feature (CPU) enables a user to answer an unspecified ringing line within a group.

The Directed Call Pickup (DPN) feature enables a user to answer a specific ringing line.

The Directed Call Pickup with Barge-in (DPU) feature enables a user to answer a specific ringing line, or barge-in a call if the call is stable and if the barged-in extension subscribes to the Directed Call Pickup with Barge-in termination feature.

In this document, a Pickup Group is defined as a group of users able to pick up each other's ringing calls using the Group Pickup Feature. A user may optionally belong to one pickup group. A user is not permitted to belong to multiple pickup groups. All users within a pickup group must belong to the same Business Group. These restrictions are documented in [GR 590].

The Group Call Pickup (CPU) feature applies within a Call Pickup Group. All extensions in the Pickup group subscribe to the CPU feature.

The Directed Call Pickup (DPN) feature enables a user to answer a specific ringing line within the Business group. An extension subscribes to the Directed Call Pickup feature.

The Directed Call Pickup with Barge-in (DPU) feature is identical to the Directed Call Pickup feature (DPN) when an incoming call is ringing. When there is a stable call instead of a ringing call, the user will barge-in the call if the barged-in extension subscribes to the DPU termination feature. An extension subscribes independently to the access and/or termination aspects of the DPU feature.

There are three separate feature access codes, for the three features.

For UEs that do not have an analog phone interface, the analog user interface requirements of Telcordia [GR 590] may be replaced with user interactions that provide similar results such as pressing a button on the device or clicking a virtual button on a screen. When an extension number must be specified, such a UE may allow the extension number to be entered either on a phone keypad or in a device-specific manner. An optional visual indication of calls that may be picked-up may be available.

6.16.5.1 Group Call Pickup (CPU)

A UE with the Call Pickup feature enables a user to answer any ringing line within their predefined pick-up group. A pick-up group is a Business Group Administrator-defined set of users, to which the Call Pickup feature applies. To pick up a ringing call, a user dials the Call Pickup feature access code. The user is then connected to the caller. If more than one line in the pick-up group is ringing, the call that has been ringing the longest is answered.

The Call Pickup attempt may fail due to terminating restrictions assigned on the requesting UE or because there are no ringing calls.

6.16.5.2 Directed Call Pickup (DPN)

A UE with the Directed Call Pickup feature enables a user to answer a call directed to another phone by dialing the respective feature activation code followed by the extension of the ringing phone.

The Directed Call Pickup attempt may fail due to terminating restrictions assigned on the requesting UE or because there is no ringing call at the specified extension. When a UE is not allowed to receive external calls (terminating restriction) it will not be allowed to pick up external calls.

6.16.5.3 Directed Call Pickup with Barge-In (DPU)

In addition to the ability to pick up a call directed to another user, this version of the Directed Call Pick-Up service enables the user to barge-in on the call if already answered by a line that subscribes to the DPU termination feature, thereby creating a three-way call. An optional warning/notification tone may be played when a barge-in occurs. Note that legal requirements may render the notification tone mandatory in certain locations. If the user being barged-in disconnects, a two-way call is established with the third-party. The user being barged-in is the controller of the three-way conference and has the option of dropping the third-party as in a regular three-way call (e.g., by flash-hook for a UE that supports an analog phone interface).

The Directed Call Pickup with Barge-in attempt may fail due to terminating restrictions assigned on the requesting UE or because there is no ringing or established call at the specified extension or because the specified extension does not have the DPU termination feature or because the barged-in user is the controller of a multiway call.

6.16.5.4 Feature Interactions

No barge-in on multi-party calls when the Target extension is the focus of an existing multi-party call, as above.

Originating features execute on Call Pickup attempt.

6.16.6 Outbound Call Restriction (Calling Plan Outgoing)

The Outbound Call Restriction (Calling Plan Outgoing) (OCR) feature provides selective Class of Service (COS) restrictions based on administrator provisioning. OCR configuration is available to Business Group Administrators via the Web Portal interface. As a group feature, OCR is typically applied to a group of users, but no restrictions are placed against configuring this feature on an individual user basis.

The administrator creates the Outbound Calling Plan by selecting particular Class of Service restrictions, and then applies these to certain users or groups. The COS restrictions can include:

- Toll calling restrictions

  Users will be permitted to make local calls only

- International calling restrictions (011+)

  Users will only be permitted to make calls that terminate within the country

- Dial around (101XXXX Casual calling)

  Users will not be allowed to make 101XXXX calls.

- Operator assisted calls (0+)

  Operator-Request Calls (0-; i.e., 0 followed by # or 4-second timeout)

- Premium calls (900)

  Users will not be allowed to dial 1-900-XXX-XXXX

- Black list

  The Business Group Administrator can configure a list of NPA, NXX, or NPA-NXX-XXXX numbers to which outgoing calls will be blocked.

- Forward/Transfer

By enabling this switch, the Administrator can prohibit the user or user group from forwarding or transferring calls to any prohibited number contained in the Outbound Calling Plan. This is a Phase 2 requirement.

Service providers may choose to define some of these restrictions according to the geographical area they will be covering (e.g., Intrastate call restrictions may not make sense to a provider that straddles a state border.)

- Default Conditions
  - If an Outbound Calling Plan has not been assigned to a user or user group, no calling restrictions will be enforced.
- Outbound Call Restriction Overrides

  The Business Group Administrator can also associate Outbound Calling Plans with Authorization Codes to allow for varying levels of call restriction. For example, User A is part of a group which is not authorized to make international calls. When User A dials an international number he is routed to an IVR system which could prompt him to enter an Authorization Code. If the code he enters maps to an Outbound Calling Plan that does not restrict international calls, the call will complete. This is a Phase 2 feature requirement.
- Restricted Call Treatment

  When a user dials a number that is restricted by his assigned Outbound Calling Plan, the call shall be routed to an IVR system for playback of a restriction announcement, or reorder tone shall be applied. The user may also be presented with the option to enter an Authorization Code at this time to override the restriction.
- Exemption List:

  It may be desirable to add an exemption list to the Outbound Call Restriction feature (or an Exemption List (White List) feature could be added independently.) Certain emergency numbers (e.g., 9-1-1, local fire/EMS) could be added to the list, thereby preventing these calls from being inadvertently blocked due to an improperly configured calling plan.

The Telcordia reference is [GR 564] Code Restriction and Diversion.

6.16.6.1 Feature Interactions

Activation of call forwarding variable should not be permitted to a prohibited destination. The BSS AS gives special service error treatment in response to such an activation request.

Emergency numbers (e.g., "911") should not be accepted in a list of prohibited destinations.

6.16.7 Charge Number Service

Charge Number Service allows Business Group Administrators to associate a charge number with an individual user or group. The usage of the charge number service is transparent to the caller. Users cannot activate/deactivate the charge number service via VSC or through an IVR prompt. The charge number service applies only to outgoing calls. Upon call origination, the Application Server inserts the associated charge number in the outgoing INVITE (or other SIP method/header - the particular mechanism is currently undefined). The charge number is then included in the user’s call detail records to be processed by the SMBs billing applications or used in the PSTN network.

6.16.7.1 Feature Interactions

The charge number can only be configured by a group member with "Admin" privileges. Entering an Authorization Code will not override the charge number or cause the charge number to be omitted from call signaling or CDRs.

6.16.8 Loudspeaker Paging

The loudspeaker paging feature allows a caller to simultaneously address a group of users within a Business Group via a unidirectional voice path, i.e., a page. To initiate a page, the caller dials a feature access number or VSC followed by a number to specify a Paging Group. A Paging Group consists of user identities provisioned by the Business Group Administrator and is identified by one or more unique digits. The loudspeaker paging feature is only applicable to UEs with auto-answer and loudspeaker capabilities. Integration with legacy paging systems and talkback abilities are deemed out-of-scope. The Business Group Administrator will provision Paging Groups via the Web Portal interface.

To initiate a page, the caller either:

- dials a feature access number or VSC code followed by the group number to which the page will be directed. or
- direct dials a private numbering plan number or speed dial number which maps to the desired paging group.

The particulars of the numbering plan will be left to the SMB. The Business Group Administrator will have privileges to configure the paging groups via the Web Portal interface.

Synchronization issues - Inter-stream synchronization is desirable in the case of loudspeaker paging where feature phones receiving a paging announcement will often be in close proximity. A preferred method of synchronization (e.g., delay playout using an RTCP RTT algorithm) should be specified.

6.16.8.1 Feature Interactions

Users with the Do Not Disturb feature enabled will not receive a Loudspeaker Page.

The loudspeaker paging feature will override all other configured terminating call features such as:

- Call forwarding (and variants) - Paging requests will not be forwarded to numbers that are not contained in the paging group
- Shared call appearance - Only the primary number associated with an endpoint will be paged
- Sequential ring
- Simultaneous ring
- Hunt groups - pilot numbers are not allowable in the Paging Group
- Call waiting - No audible or visual notification will be given when a paged line is busy. If notification is desired however, this should be configurable (e.g., by using the "Priv-Answer-Mode" header to differentiate an important announcement)

6.17 Web Portal Services

Web portals are an increasingly important aspect of all kinds of voice communications services. They provide a means for subscribers to interact with their voice services via a user interface that offers far greater richness of function than their telephone instruments, and that is accessible from any location with connectivity to the Internet.

Web portals can deliver value to subscribers and to service providers in the following ways:

- They provide an alternative means for subscribers to modify the settings of voice services such as Call Forwarding Variable that is typically easier to use and more intuitive than the traditional user interface to such services - via entry of a digit string commencing with a Vertical Service Code at a telephone. They also enable service extensions that require the user to supply alpha-numeric information (e.g., a non-numeric SIP URI) which is otherwise difficult to accomplish using the traditional telephone user interface.

- They enable services that depend on complex configuration such as Automated Attendant to be set up and managed by subscribers themselves, where traditionally it is has been necessary for service providers to establish and maintain the service configuration on behalf of subscribers.
- They provide a means for subscribers to access and maintain a variety of additional data that is associated with or supports their use of Business SIP Services, including Call Logs and Address Books.
- They provide a richer user interface than the traditional telephone instrument for making outbound calls, handling inbound calls, manipulating call appearances, and invoking subscriber-controlled features such as Call Hold and Call Transfer. These functions are referred to collectively as Third Party Call Control.
- They provide a means for subscribers to order additional voice services and calling features, whereby new services can be enabled immediately and automatically following an online request made via a Web portal. This reduces the cost to the service provider of taking and fulfilling orders for new services and reduces the lead time to make the new services available to the subscriber.

This section describes the function of Web portals as it relates to the first four items above. Support for the ordering of additional voice services and calling features is considered out of scope of this document.

A subscriber is able to access the Web portal using standard web browsers running on commonly used desktop operating systems such as Microsoft Windows and Apple Mac. It should not be necessary for the subscriber to download and install proprietary plug-ins for the purpose of using any of the services available from the Web portal.

6.17.1 User Sign-on

When a user points his web browser at the service provider's Web portal, the Web portal authenticates the user. Typically this is achieved by prompting the user to provide a username and password, although other sign-on technologies are not precluded, such as certificates or a biometric fingerprint scanner attached to the user’s computer. Where password-based authentication is used, it should be possible for a user to choose a memorable password whilst satisfying operator-definable policy on password strength. The user may also change the password at any time, for example if they believe that it may have been compromised.

Once signed on, the user is presented with an initial page from which he is able to access all the services to which he has subscribed. A well designed user interface will allow the most frequently-used functions to be accessed directly from the initial page, with even the most rarely-used feature being accessible via a small number of mouse clicks away from this page. The initial page may also draw the user’s attention to the status of some key features such as Call Forwarding Variable or the number of unheard messages in the subscriber's voicemail inbox.

Following sign-on and user authentication, the Web portal is expected to provide each user with an integrated view of all of the services to which he is subscribed. The fact that these services may actually be delivered by multiple distinct back-end call processing systems should not be visible to the user. For example, it would be common practice for voice messaging and call forwarding services to be implemented by different Application Servers, perhaps provided by different vendors, but the user might be presented with a single Web page that shows the status of both the subscriber's voicemail inbox and the current setting of his Call Forwarding Variable service.

To reduce the risk of an unauthorized person accessing the user’s services, a user’s session will time out after a period of inactivity. A user may also manually log out of the Web portal.

6.17.1.1 User Categories and Privileges

Different individual users within a given SMB organization have access to different sets of Web portal capabilities, reflecting their specific roles in the management of voice services for that SMB.

Two main classes of users are defined: "Business Group Administrators" and "Individual Users".

The exact definition of the privileges and capabilities available to each class of user is for the service provider to determine, but typically each class of user would be provided with the following abilities.

An Individual User can use the Web portal to control and configure the calling features and services that are provided on the line designated for his individual use, including the following:

- Call Forwarding - No Answer, Busy, Selective, Variable
- Selective Call Acceptance / Rejection
- Do Not Disturb
- Simultaneous Ring
- Sequential Ring
- Distinctive Ringing
- Call Push / Call Pull configuration
- Speed Dial
- Remote Office

A Business Group Administrator is typically a nominated employee of the business who is empowered to administer calling features and services for the Business Group as a whole, in addition to those assigned to the line designated for his individual use. A larger Business Group consisting of several departments may have some Administrators with permissions to only manage a subset of the Business Group (e.g., all lines within a department). In this case, a top level Administrator is able to manage the administrative hierarchy via the Web portal.

An Administrator can use the Web portal to manage calling features and services that apply to the Business Group as a whole, or to some subset of lines that serve the SMB, including for example the following:

- Mapping between Extension Dialing codes and DID numbers
- Account Codes
- Authorization Codes
- Hunt Groups
- Call Pickup Groups
- Inbound and Outbound Call Restrictions
- Call Intercept
- Configurable Feature Code Prefix
- Automated Attendant
- Media On Hold

A service provider may choose to make available to Administrators via the Web portal some management functions which would traditionally have been performed by a customer services representative, such as:

- Resetting an Individual User's forgotten password.
- Re-initializing a line's services, for example when an employee leaves the SMB and is replaced by a new employee. This operation disables services such as Call Forwarding Variable and resets personal settings such as passwords, speed dials and voicemail greetings.
- Barge-in privileges.
- Self-subscription to new services for the group or specific lines.

- Assignment of functions to hard and soft keys available on a SIP phone.

6.17.2 Call Logs and Address Book

The Web portal provides facilities for a user to view historical data about calls that he has made and received, in the form of an itemized list showing the identity of the remote party, the time and date of the call and the duration of the call. A Business Group Administrator has access to consolidated Call Logs at the level of the Business Group, where he can view details of calls for the Business Group as a whole in chronological order, or grouped by extension number / DID number.

The Web portal provides facilities for a user to maintain a network-based Address Book, in which he can store details of business or personal contacts with phone numbers. Data stored in the Address Book supports a number of other functions offered by the Web portal and by Third Party Call Control applications, for example:

- Entries in Call Logs are annotated with names that are obtained by looking up the number of the remote party in the Address book.
- Web pages that provide facilities for editing screening lists in support of features such as Selective Call Forwarding display screening lists annotated with names looked up in the Address Book, and can offer a means to select entries from the Address Book for inclusion in screening lists.
- The calling name displayed on a SIP business phone, analog calling name display or incoming call notification function of a Third Party Call Control application can be derived by looking up the number of the calling party in the Address Book.
- Click-to-dial calls may be initiated quickly and conveniently by selecting an entry in the Address Book.

The Web portal may also provide facilities to enable a Business Group Administrator to maintain a shared Address Book that includes details of contacts which are of interest to the SMB as a whole. Individual Users have read-only access to the shared Address Book.

6.17.3 Appearance of User Interface

The Web portal for Business SIP Services appears to the user as seamlessly integrated part of the overall Web experience that is made available by the service provider, which may include a range of other functions that are outside the scope of Business SIP Services, for example financial details of the user's account with the service provider. All of these functions are available to the user following a single sign-on and user authentication process.

The Web portal for Business SIP Services typically shares the same branding, in terms of logos, color schemes, graphical elements and service naming terminology as the rest of the service provider's Web site.

The Web portal user interface may allow the user to specify display preferences which persist between sessions and between locations from which the user accesses the Web portal. For example, these settings could determine the features which appear on the user’s main page, how date and phone numbers are formatted and which language the user interface is displayed in.

6.17.4 Feature Interactions

Many BSS features are suitable for control using a Web Portal.

6.18 Third Party Call Control Features

Third Party Call Control refers to the use of a personal computer to provide graphical user interface functions for initiating calls from or interacting directly with calls received at telephones served by Business SIP Services. The graphical user interface provides a rich and user-friendly means to control calls as an alternative to the use of flash hook and Vertical Service Codes available on a traditional phone. The user interface may be provided by means of a standalone application that displays its own window, a Web application supported by a standard browser, or a "plug-in" module integrated with other some other desktop application such as Microsoft Outlook.

Third Party Call Control comprises a group of capabilities that are typically combined within a single application. In general terms, these capabilities can be described under three distinct categories:

- Support for making calls - Click-to-dial.
- Support for handling incoming calls - Incoming Call Notification and Control.
- Support for viewing and manipulating the status of active and held calls - Call Hold/Retrieve, Call Transfer, Conferencing, Release.

Features which provide detailed and general monitoring of the user's phone state are outside the scope of Third Party Call Control, such as an attendant console application which shows the phone's current hook status.

This feature is implemented by the BSS AS and places minimal requirements on the UE. It does not depend upon direct communications between personal computers and the UE, allowing these to be located on different network segments. The communications path between the personal computer and the BSS AS is out of scope.

6.18.1 General characteristics

Third Party Call Control is applicable to users who are equipped with any type of telephone instrument, including analog handsets, SIP business phones and SIP soft phones. It does not depend upon communication between personal computers and intelligent handset devices.

The Third Party Call Control features are available when the user's personal computer has network connectivity typically found at a SMB or another remote location (when using the Remote Office feature). As a guide, any computer able to access the Business SIP Services Web portal is also able to use the Third Party Call Control features. Some specific considerations include:

- The computer may be located behind a NAT/Firewall. To maintain computer security, the firewall may heavily restrict Internet access - for example by only allowing outbound access using HTTP and HTTPS protocols.
- The computer may be connected to a different network segment than the user's SIP telephone, such as a WiFi network.
- The local computer access network may have limited or no support for Quality of Service.

The ability of a user to originate and terminate calls using his phone in the conventional way is not impacted by failure of the user's computer (e.g., software crash) or the network connection.

6.18.2 Click-to-dial

Click-to-dial allows a user to initiate a call using his computer, for example by explicitly entering the number to call, or by selecting an entry from a list of recently received calls in the Call Logs, or by selecting an entry in his Address Book.

When the user initiates a Click-to-dial call, his or her own phone starts ringing immediately, typically with a distinctive ringing cadence to distinguish this call from a normal incoming call. The Caller ID presented on this first leg of the Click-to-dial call is the target destination for the call. If the user has a SIP phone that supports auto-answer, the first call leg may be answered automatically. Otherwise, the user needs to go off-hook. At this point, the target destination number is called. The call ends when the user hangs up his phone or uses the Release Call feature described below.

A user who is already in an active call may initiate a Click-to-dial call, provided that his service permits multiple call appearances to be present on his line. The existing call could be an incoming call, an outgoing call manually dialed from the phone or a call established using Click-to-dial. In these circumstances, the existing call is put on hold while the new call is established. Subsequent behavior depends on what other Third Party Call Control functions are available to the user from the application. If none, then the user is limited to controlling the two legs of the call by means of flash hook, the effect of which is determined by the type of conventional multi-party service available to the user (Call Transfer or 3-Way Calling).

Click-to-dial calls are subject to the same BSS features as would apply if the call had been manually originated using the phone. For example, the user is unable to circumvent Outbound Call Restriction or Authorization Codes. The Click-to-dial user interface may offer the option to allow a user to enter an Account or Authorization code as an alternative to entering it using the phone's keypad.

With the exception of the Remote Office Click-to-dial service, terminating BSS features are not applied to the initial leg of Click-to-dial calls. Specifically:

- A user is able to originate Click-to-dial calls when deflection/rejection services, such as Do Not Disturb, Call Forwarding and Selective Call Acceptance, would prevent normal incoming calls from ringing the user's phone.
- The call is always presented to the user's phone even when features, such as Shared Call Appearance or Simultaneous Ring, would normally result in incoming calls ringing other phones. This will, for example, prevent an assistant's phone ringing each time a manager originates a Click-to-dial call. Where several phones share a common public user identity or phone number, there shall be a mechanism which allows the user to select which phone is to be used for Click-to-dial calls.
- The initial call leg does not appear in a user's call lists.

6.18.2.1 Feature Interactions

With the exception of Remote Office (see Section 6.15.11), a subscriber's normal terminating call services are overridden on the first call leg of a click-to-dial call. For example:

- Call forwarding (and variants) - the call is not forwarded
- Sequential ring - only the subscriber's UE rings.
- Simultaneous ring - only the subscriber's UE rings.
- Call rejection (and variants) - the call is not rejected.

A subscriber's originating call services (e.g., account codes, outbound call blocking) do apply to the second call leg to the click-to-dial target, as though the subscriber had originated the call normally from their phone. This prevents, for example, a subscriber from making click-to-dial calls to a destination which is blocked.

It should not be possible for a user to initiate an emergency call using click-to-dial; in a PacketCable network, emergency calls are routed differently to normal calls and may carry UE location information. A user should instead initiate an emergency call using their handset.

From a signaling perspective, the call leg to the click-to-dial user can appear to be a "terminating" call. When used in conjunction with Remote Office, this may interact poorly with a feature commonly found in PSTN networks where the call is released some time after the terminating party places their phone "on hook" (see Section 5 of [GR 505] and Section 13 of [GR 506]). [GR 506] indicates that the call is released between 10 and 12 seconds after a terminating party with an analog phone goes into the on-hook state (the "Timed Release Interval"). A PSTN switch compliant with [GR 505] will immediately signal the on-hook event without releasing a call, for example using the SUS (suspend) message over ISUP. However a BSS AS may not receive the suspend event because:

- There is no mechanism for signaling suspend events over SIP. The PacketCable MGC (SIP-PSTN gateway) may immediately release the call if it receives the suspend event. This will result in desirable behavior where the click-to-dial call is released.
- It is thought that some widely deployed legacy PSTN switches are not compliant to [GR 505] and don't signal anything when the phone goes on-hook. This behavior is outside the control of the PacketCable network and cannot be worked around. The call will be released once the Timed Release Interval expires. Alternatively the user can release the call using the 3PCC Release Call feature.

6.18.3 Incoming Call Notification and Control

When the user receives an incoming call, the Third Party Call Control application displays details of the call on the user's computer. The user may be currently engaged in some other activity on his computer, such as editing a spreadsheet, so the application pops up a display window in a convenient location on the screen on top of any other application that may be running. Information displayed includes:

- The calling number.
- The name of the caller. This could be obtained by querying a network-based public database or the user's own Address Book.
- Whether the user is being called directly or through a Hunt Group.

The user may answer the incoming call in the conventional way, by just going off hook. Alternatively the user may select one of the incoming call control options offered by the application, which include:

- Deflect the call to another line or to voicemail.
- Reject the call, so the caller hears an announcement.
- Answer call waiting. This option is only presented if the user is already engaged on another call. If this option is selected, the existing call is placed on hold and the new incoming call is answered.
- Answer call to headset / speaker. This option is only presented if the user has a telephone instrument that can be signaled to auto-answer to headset or loudspeaker.

6.18.3.1 Feature Interactions

The call notification will typically only display at times when a call results in the user's phone ringing (e.g., it won't be shown if the call is rejected). The call control aspect of the feature should not allow the user to circumvent service restrictions, for example by allowing a call to be forwarded to a blocked destination.

6.18.4 Call Hold/Retrieve

This feature allows the user to place an active call on hold for an extended period of time, or to retrieve a previously held call. Not constrained by the [GR 579] user interface requirements for analog phones, the graphical user interface may allow multiple calls to be held simultaneously even if the user has an analog handset.

The Third Party Call Control application displays details of each call that is present on the user's line, showing the status of each (connected or held). The user may select any held call and make it active, at which point the currently connected call is put on hold. The user may also select the currently active call, put it on hold, and then start another call. The call status display is dynamically refreshed by the application to show the current state of all calls on the user's line as new calls are added, and as calls are dropped (for example, when the remote party hangs up).

6.18.4.1 Feature Interactions

3PCC Call Hold operates independently of local Call Hold - either feature can place the call "on hold". It isn't however possible to place a call on hold using one mechanism and retrieve it using the other mechanism.

6.18.5 Call Transfer

This feature allows the user to transfer an active or held call to a third party identified by entering an explicit number to call or by selecting an address book entry. The user can choose whether to consult with the third party (attendant/consultative call transfer) or transfer directly (blind call transfer). Once the call has been transferred, the user loses the ability to monitor or control the call.

Call Hold/Retrieve is a pre-requisite for the Call Transfer feature.

6.18.5.1 Feature Interactions

The feature interactions are similar to locally initiate Call Transfer. For example, it should not be possible to transfer a call to a blocked destination

6.18.6 Conferencing

This feature allows the user to establish a multi-party call by adding a new party or previously held call to an active call. Some versions of this feature may allow more than three parties to participate in a conference. The user interface may display details of all participants in the conference and may provide additional functions such as the ability to mute some parties.

6.18.6.1 Feature Interactions

The feature interactions are similar to locally initiated multi-party conference calling.

6.18.7 Release Call

This feature allows the user to the user to disconnect a call which has been answered. The call may be an active call, a held call, or a single party in a conference.

6.19 Email, Address Book and Directory Integration

6.19.1 Overview

The integration of PacketCable services with the user's desktop will allow a user to combine the capabilities of a soft phone or other communications application and their Email, local address book and network directory services. This enables flows in which:

- An email application triggers a soft phone to initiate an outgoing call.
- A soft phone queries a local address book and remote directory to retrieve information to enhance the presentation of an incoming call.
- An email application enables the "Click to Dial" call on the "Hard" SIP Phone.

The integration of the Email client (e.g., Outlook, Notes) with the local soft phone enables the 1PCC (1^{st} Party Call Control) whereas integration with 3PCC services enables the control of the Hard SIP Phone.

Using the same approach, the integration can be extended beyond the Email client to include other applications such as document editors (e.g., Word, Excel) so that a user can place a call to a contact found in a document.

In the desktop environment, existing tightly integrated products provide most functions in a proprietary manner. Specific behaviors depend heavily on the specific operating system and applications used. Replacing these deeply rooted products would prove difficult. In additional, the desktop environment is constantly evolving and changing. Therefore, no formal specification of the desktop integration applications is to be done in the BSS specification.

Instead, this specification permits proprietary solutions within the desktop. Integration with network services should incorporate standard capabilities. Protocols that support directory access include LDAP [RFC 4510], and derivative protocols such as [DSML v2]. Protocols that support application messaging include [SOAP].

The following sections present the flows that should be supported by the desktop integration of PacketCable services.

6.19.2 First Party Outgoing Call

In this scenario, the address book is integrated with a desktop application or "plug-in" to enable the user to make calls by clicking on the contact embedded in a Document or Email. The desktop application, or a "plug-in", accesses the local address book or corporate directory or public network directory to get details about the called party and then delegates the "call making" functionality to the Soft Phone.

When the user clicks on a contact in the Email (Outlook/Notes/etc.) or document (Word/Excel/etc.), the Desktop application queries the local address book and then a corporate directory or the service provider directory. The query to the directories may, for example, translates the Outlook Contact info for the called party to the SIP URI. It may also figure out additional information, including whether the call is allowed.

The Desktop application then triggers the local soft phone and passes on the details for making a call. The 3PCC application sets up the call between the user and the Called Party.

6.19.3 First Party Incoming Call

In this scenario, the soft phone is integrated with an address book to enhance the presentation of the incoming call.

The soft phone accesses the local address book and a private or public network directory to get details about the calling party as it completes an incoming call.

With the additional information, the subscriber will better know whether to accept or reject an incoming call. The system can improve over time, since as unsolicited calls are received, the subscriber can place contacts in a local or corporate reject list so that future calls from the calling party can be rejected.

When an incoming call is received on the soft phone, it invokes a Desktop application or plug-in which queries the Address Book or Directory. The information is returned to the soft phone for display.

6.19.4 Third Party Click to Dial Call

In this scenario, the address book is again integrated with a desktop application to enable the user to make calls by clicking on the contact embedded in a Document or Email. In this case, the returned information is passed out to a 3PCC application to be used in a call, e.g., a "Click-To-Dial" (CTD) call.

When the user clicks on a contact in the Outlook/ Notes/Word/Excel/etc., the Desktop application queries the local address book and a private or public directory. As in the 1PCC outgoing flow, the query may return the SIP URI of the Called Party.

The Desktop application then triggers the remote 3PCC application and passes on the details for making a call. The 3PCC application sets up the call between the hard phone and the Called Party.

6.20 Security and Authentication

The security specifications applicable to Business SIP Services are provided by the documents [PKT 33.203] and [PKT 24.229].

At a minimum devices providing support for Business SIP Services need to provide support for Digest based authentication as described in the above references. Digest authentication can be applied to any SIP requests for which the network is configured to require authentication. Support should also be provided for security mechanisms to provide appropriate levels of encryption for both signaling and media for nomadic clients that support Business SIP Services.

Further a UE providing support for Business SIP Services can only provide such service if it is considered to be in an in-service state. That is, dial-tone or other appropriate indication that the device is ready to initiate a call attempt is applied only if the UE is successfully registered with the core network and it has verified that it has network connectivity that enables it to initiate a session.

6.21 Device Provisioning

The device provisioning of Embedded User Equipment and Non-Embedded User Equipment endpoints is expected to use the PacketCable mechanisms defined for these types of devices.

6.22 Device Requirements

One of the potential BSS devices is an embedded UE device that is similar to the E-DVA defined for the PacketCable RST application. In particular, this BSS device follows closely the RST Feature Specification [RSTF] for its execution of telephony features, with the necessary modifications and enhancements for the BSS application environment. Furthermore, such device follows closely the E-DVA Specification [RST E-DVA] for the requirements on analog line interface and other device-specific functionality, with the necessary extensions defined for the BSS. Such extensions can potentially include the increased maximum number of analog line interfaces, prolonged battery back-up time, and support for wideband codec, etc.

Other potential BSS devices include desktop SIP phones and WiFi SIP phones. In principle, these devices conform to the PacketCable RST Feature Specification [RSTF] for their execution of telephony features, with the necessary modifications and enhancements for the BSS and for their specific user interfaces. Soft SIP phones may also be deployed for the BSS application.

6.23 NAT/Firewall Traversal

Enterprise NAT/Firewall traversal will be supported using the NAT traversal procedures defined in PacketCable. Enterprises may need to adjust their Firewall policies to allow these procedures to work.

7 ARCHITECTURE

7.1 Web Portal

The figure below illustrates the Web Portal reference architecture.

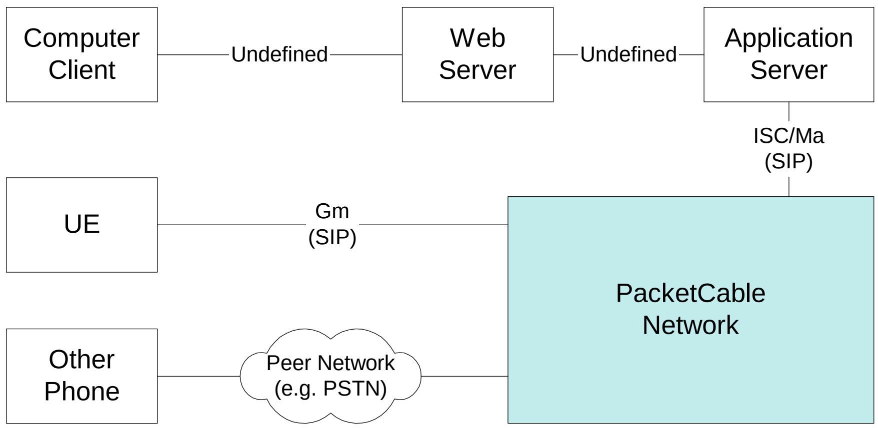

Figure 2 - Web Portal reference architecture

A client application, such as a web browser, runs on the personal computer and interfaces with one or more Web Servers located in the operator's network. The path between the personal computer and the Web Server may contain NAT/Firewalls and could be routed over networks outside of the operator's control (e.g., the Internet). Web browsers utilize HTTP and HTTPS for transport, although other protocols could be used by proprietary applications and browser plug-ins (e.g., Adobe Flash). The use of HTTPS is recommended since it provides data security and successfully traverses NAT/firewalls.

A client running on a personal computer accesses BSS Web applications through a Web Server (WS), typically using the HTTP or HTTPS protocols. The WS represents the trust boundary between untrusted parts of the network (Local/Access network) and trusted parts of the network (Core, Application). The functions provided by the WS include:

- Authenticating the user.
- Supplying the user interface in a form which a web browser client can render. This may include supplying the browser with static content (e.g., style sheets, images etc.) and dynamically generated content (e.g., a user's settings).
- Accepting requests from the client to invoke an application or service.

The Web Server (WS) accesses user services through one or more BSS Application Servers. A WS may also access other back-end systems, such as a billing or email system, the details of which are outside the scope of this document. It is increasingly common for the interface between a WS and its back-end application to be based upon W3C's Web Services Architecture (utilizing SOAP for message passing and WSDL for interface definition). The precise details of the WS-AS interface are undefined in BSS.

The rationale for keeping the AS-WS interface details outside BSS scope is that for many Web Portal services (e.g., accessing call service settings), it would be identical/similar to the OSS-AS interface. The OSS-AS interface is outside BSS scope.

7.1.1 User Authentication

7.1.1.1 User Credentials

For UE authentication, the credentials are stored persistently within the UE or automatically obtained during bootstrapping as part of device provisioning or a one-time user operation (e.g., when first configuring a SIP soft phone). The user will not normally be aware of the underlying technology used to authenticate the UE. Contrastingly, Web Portal authentication will usually prompt the user to enter a username and password each time they use the service, which is why it is important that the user is able to change their password to a memorable value. This means that the Web Portal password needs to be independent of any UE authentication password - indeed a user with multiple UEs could have multiple independent passwords (if each UE has its own Private User Identity).

The nature of the user's Web Portal password means that, like other BSS application data, it does not need to be stored as part of the core IMS data in the HSS (although a Web Portal implementation is permitted to store it as transparent HSS data). Web portal user authentication therefore does not place any new requirements on the PacketCable core.

7.1.1.2 Web Single Sign-On

An operator's Web Portal may actually be realized using a number of web servers, each providing access to different applications. It is required that each WS is able to confirm the identity of the user without requiring that the user manually re-authenticates themselves each time they access a different application.

There are several Web Single Sign-On technologies in use today, with many industry standards bodies adopting OASIS [SAML]. They all operate on the same underlying principle where a user is authenticated on one WS (the "Identity Provider") which supplies a token identifying the identity of the user. The client presents this token to another WS (the "Service Provider"). The Service Provider is able to verify user's identity from a digitally signed assertion within the token or through direct communications with the Identity Provider. A number of mechanisms exist for a web browser to present the token to Service Provider WS, including embedding it within the HTTP GET URL or within HTTP POST data.

7.2 Third Party Call Control (3PCC)

Third Party Call Control (3PCC) is one Web Portal application defined in BSS. The 3PCC AS is responsible for converting between the 3PCC primitives used by the 3PCC WS and SIP used by the PacketCable Core network (ISC/Ma reference points). The details of the interface between the 3PCC WS and 3PCC AS is undefined in BSS, it may be based upon one of a number of existing 3PCC control protocols such as [CSTA] or [Parlay X].

The BSS specification describes details of the 3PCC SIP flows. The flows for the Remote Office feature should minimize the number of new interoperability requirements placed onto non-BSS SIP devices such as MGCs, PacketCable 1.x CMSs and RST E-DVAs. This would tend to favor an implementation where the 3PCC AS manipulates call legs as a SIP Back-to-Back User Agent (B2BUA), for example using the flows defined in [RFC 3725].

SIP does not currently specify a mechanism to allow a third party to control certain features locally implemented on the UE. For example, a call placed on local hold using the UE's user interface (e.g., flash hook on an analog phone) cannot be removed from hold by the 3PCC AS using standard SIP. Such limitations are inherent for Remote Office phones located in the PSTN, as there is no way to control these features over standard PSTN signaling protocols (e.g., ISUP).

7.3 Email, Address Book and Directory Integration

The figure below is the reference model for the interactions with desktop software.

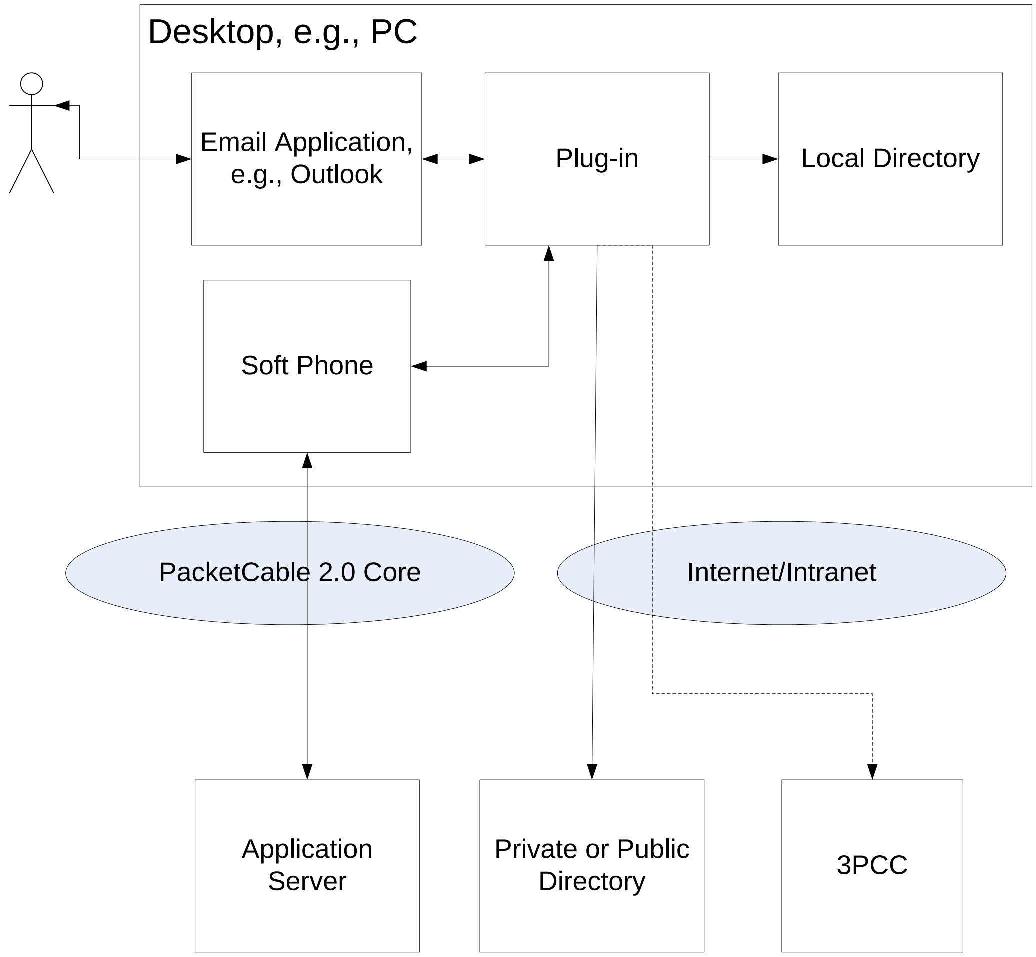

Figure 3 - Desktop software interactions for email, Address Book and Directory Integration

Authentication may be required for the remote directory or 3PCC application access. Local desktop functions will require no authentication beyond the user's initial authentication with the desktop.

7.4 Call Push, Call Pull (Call Jump)

The deployment issues discussed can be avoided if Call Jump is prohibited for calls originated on the cellular device, with cellular network Call Transfer being used in such instances. This however may lead to confusion with the user as the feature experience between Call Transfer and Call Jump is not equivalent.

The aforementioned restriction on Call Jump to Cable MSO only devices may be alleviated should a specialized client be provided on the cellular device. The main need for a specialized client is the simplification of the user experience. In case the Call Jump is invoked from the cellular device complex user interactions such as Two Stage Dialing procedures are needed to convey Call Jump signaling information to the Cable MSO Network. This can be mitigated by a specialized client.

A specialized client for cellular devices however leads to issues of device/vendor selection and potentially porting the client to various handset Operating Systems.

Note also that use of mechanisms such as two-stage dialing (which is the only option entirely under the control of the Cable MSO) even if mitigated by a specialized client still leads to user experience issues as this dialing scheme is entirely independent of and cannot be used in conjunction with native schemes available on the cellular device such as dialing from the address book or call logs.

Appendix I User Interface Tones and Signals

PacketCable Business SIP Services Features	PC 1.x Reference	Description	Additional Info
	0-9,*,#,A, B,C,D	MFPB (DTMF) tones	
	Bz	Busy tone	Default time-out = 30 s
	Cf	Confirmation tone	
Calling Number Delivery Calling Name Delivery Calling Identity Delivery Blocking	ci (ti, nu, na)	Caller Id	"ti" denotes time, "nu" denotes number, and "na" denotes name
	Dl	Dial tone	Time-out = 16 s
Fax Support	Ft	Fax tone	PC 1.5 T.38 and/or V.152
Modem Support	Mt	Modem tones	PC 1.5 T.38 and/or V.152
Audible Message Waiting Indicator	Mwi	Message waiting indicator	Time-out = 16 s
	Ot	Off-hook warning tone (ROH)	Time-out = configurable
Distinctive Ringing	r0, r1, r2, r3, r4, r5, r6 or r7	Distinctive ringing (0..7)	Time-out = configurable
	Rg	Ringing	Time-out = 180 s
	Ro	Reorder tone	Time-out = 30 s
Visual/Audible indication of Call Diversion (ring splash when calls forwarded)	Rs	Ringsplash	
	Rt	Ring back tone	Time-out = 180 s
	Sl	Stutter dial tone	Time-out = 16 s
Fax, Modem, TDD call support	TDD	Telecomm Devices for the Deaf (TDD) tones	PC 1.5 T.38 and/or V.152
Visual Message Waiting Indicator	Vmwi	Visual message waiting indicator	
Call Waiting Cancel Call Waiting	wt1, wt2, wt3, wt4	Call waiting tones	Time-out = 12 s
	X	MFPB (DTMF) tones wildcard	Matches any of the digits "0-9"
	Ta	Alerting Signal Tone	
	Td	Special Dial Tone	
	Ti	Special Info Tone	
	Tr	Release Tone	
	Tc	Congestion Tone	

The tone definitions are as follows:

Busy tone (bz): Busy tone is used to indicate the called UE can not be accessed at the requested time. A SIP response of 486 Busy Here is an example of SIP response that would cause the UE to generate a busy tone. The busy tone necessary for PSTN analog line emulation is defined by the local administration, and can be re-defined via provisioning [GR 506], [EG 201 188] and [EN 300 001]. It is considered an error to try and play busy tone on a phone that is off line (on hook). UEs may support provisioned text or graphical indication of a busy call progress state.

Confirmation tone (cf): Confirmation tone is used to inform the user that a feature has been activated or deactivated. Confirmation Tone is defined by the local administration, and can be re-defined via provisioning [GR 506], [EG 201 188] and [EN 300 001]. It is considered an error to try and play confirmation tone on a phone that is off line (on hook). UEs can support provisioned text or graphical indication of a confirmation call progress state.

Fax tone (ft): The fax tone event is generated whenever a fax call is detected by presence of V.21 fax preamble or T.30 CNG tone (if provisioned). See [CODEC-MEDIA], [ITU-T T.30] and [ITU-T V.21].

Modem tones (mt): The modem tone event is generated whenever a data call is detected by presence of V.25 answer tone (ANS) with or without phase reversal or V.8 modified answer tone (ANSam) with or without phase reversal. See [ITU-T V.25] and [ITU-T V.8].

Dial-tone (dl): Dial tone indicates a UE is ready to receive address information. Dial Tone is defined by the local administration, and can be re-defined via provisioning. See [GR 506], [EG 201 188] and [EN 300 001]. It is considered an error to try and play dial-tone on a phone that is off line (on hook). UEs can support provisioned text or graphical indication that it is ready to receive address information.

Message Waiting Indicator (mwi): Message waiting tones are used with message waiting features. An example use is for the UE to send the Message Waiting Indicator when the user goes off hook as an indication that a message is waiting. Message Waiting Indicator is defined by the local administration, and can be re-defined via provisioning. See [GR 506], [EG 201 188] and [EN 300 001]. It is considered an error to try and play message waiting indicator on a phone that is off line (on hook).

Off-hook warning tone (ot): This tone is typically used to indicate the UE is executing the procedure to place the line in a permanent signal state. The purpose of the signal is to notify the user to place the line in the on hook state. Receiver Off Hook Tone (ROH Tone) or "howler" tone is defined by the local administration, and can be re-defined via provisioning. See [GR 506], [EG 201 188] and [EN 300 001]. It is considered an error to try and play off-hook warning tone on a phone that is off line (on hook).

Reorder tone (ro): Re-order tone is typically used to indicate the UE or network facilities needed to make the call are not available. Re-order tone is defined by the local administration, and can be re-defined via provisioning. See [GR 506], [EG 201 188] and [EN 300 001]. It is considered an error to try and play reorder tone on a phone that is off line (on hook).

Ring back tone (rt): Audible ring tone is typically used to notify the user that the called address is being alerted. Audible Ring Tone is defined by the local administration, and can be re-defined via provisioning. See [GR 506], [EG 201 188] and [EN 300 001]. The ring back signal can be applied to both an endpoint and a connection. When the ring back signal is applied to an endpoint, it is considered an error to try and play ring back tones, if the endpoint is considered off line (on hook).

Stutter Dial tone (sl): Stutter Dial Tone (also called Recall Dial Tone) is defined by the local administration, and can be re-defined via provisioning. See [GR 506], [EG 201 188] and [EN 300 001]. The stutter dial tone signal may be parameterized with the signal parameter "del" which specifies a delay in ms to apply between the confirmation tone and the dial tone. One of the features that uses this is Speed Dialing. The following applies stutter dial tone with a delay of 1.5 s between the confirmation tone and the dial tone:

S: sl(del=1500)

It is considered an error to try and play stutter dial tone on a phone that is off line (on hook).

Call Waiting tone1 (wt1, .., wt4): Call Waiting tones are defined by the local administration, and can be re-defined via provisioning. See [GR 506], [GR 571], [EG 201 188] and [EN 300 001].

Special Dial Tone (td): Special Dial tones are defined by the local administration, and can be re-defined via provisioning. See [OB 781].

Special Info Tone (ti): Special Info tones are defined by the local administration, and can be re-defined via provisioning. See [OB 781].

Release Tone (tr): Release tones are defined by the local administration, and can be re-defined via provisioning. See [OB 781].

Congestion Tone (tc): Congestion tones are defined by the local administration, and can be re-defined via provisioning. See [OB 781].

Programmable Signal (ps1, ps2, ps3, ps4): Programmable Signal tones are defined by the local administration, and can be re-defined via provisioning. See [OB 781].

MFPB (DTMF) tones (0-9, *, #, A, B, C, D): Detection and generation of MFPB (DTMF) signals is described in [GR 506] and [EN 300 001].

Caller Id (ci(time, number, name)): See [GR 506], [GR 1188], [GR 31], [EN 300 659-1], and [EN 300 659-3]. Each of the three fields is optional; however, each of the commas is always included:

The time parameter is coded as "MM/DD/HH/MM", where MM is a two-digit value for Month between 01 and 12, DD is a two-digit value for Day between 1 and 31, and Hour and Minute are two-digit values coded according to military local time, e.g., 00 is midnight, 01 is 1 a.m., and 13 is 1 p.m.
The number parameter is coded as an ASCII character string of decimal digits that identify the calling line number. White spaces are permitted if the string is quoted, however they are ignored.
The name parameter is coded as a string of ASCII characters that identify the calling line name. White spaces, commas, and parentheses are permitted if the string is quoted.

Ringsplash (rs): Ringsplash, also known as "Reminder ring" is a burst of power ringing that may be applied to the physical forwarding line (when idle) to indicate that a call has been forwarded and to remind the user that a Call Forwarding sub feature is active. This signal is defined by the local administration, and can be re-defined via provisioning. See [GR 586], [EG 201 188], and [EN 300 001].

Telecomm Devices for the Deaf tones (TDD): The TDD event is generated whenever a TDD call is detected, see [ITU-T V.18].

Visual Message Waiting Indicator (vmwi): The transmission of the VMWI messages conforms to the requirements in [GR 30], [SR-TSV-002476], [GR 1401]; [GR 506], [EN 300 659-1], and [EN 300 659-3]. VMWI messages are only sent from the embedded UE to the attached equipment when the line is idle. If new messages arrive while the line is busy, the VMWI indicator message is delayed until the line goes back to the idle state. The Call Agent should periodically refresh the CPE's visual indicator.

DTMF tones wildcard (X): The DTMF tones wildcard matches any DTMF digit between 0 and 9.

Alerting Signal Tone (ta): Alerting Signal tones are defined by the local administration, and may be re-defined via provisioning. See [OB 781].

Appendix II Vertical Service Codes

The following list provides some current sample usages of vertical service codes (star codes) in telephone networks:

*40 - Change Forward-To Number for Customer Programmable Call Forwarding Busy Line
*41 - Six-Way Conference Calling Activation
*42 - Change Forward-To Number for Customer Programmable Call Forwarding Don't Answer
*43 - Drop last member of Six-Way Conference Call
*44 - Voice Activated Dialing
*45 - Voice Dialing Extended Dial Tone
*46 - French Voice Activated Network Control
*47 - Override Feature Authorization
*48 - Override Do Not Disturb
*49 - Long Distance Signal
*50 - Voice Activated Network Control
*51 - Who Called Me?
*52 - Single Line Variety Package (SVP) - Call Hold
*53 - Single Line Variety Package (SVP) - Distinctive Ring B
*54 - Single Line Variety Package (SVP) - Distinctive Ring C
*55 - Single Line Variety Package (SVP) - Distinctive Ring D
*56 - Change Forward-To Number for ISDN Call Forwarding
*57 - Customer Originated Trace
*58 - ISDN MBKS Manual Exclusion Activation
*59 - ISDN MBKS Manual Exclusion Deactivation
*60 - Selective Call Rejection Activation
*61 - Distinctive Ringing/Call Waiting Activation
*62 - Selective Call Waiting
*63 - Selective Call Forwarding Activation
*64 - Selective Call Acceptance Activation
*65 - Calling Number Delivery Activation
*66 - Automatic Callback Activation
*67 - Caller ID Per-Call Blocking
*68 - Call Forwarding Busy Line/Don't Answer Activation
*69 - Automatic Recall Activation
*70 - Cancel Call Waiting
*71 - Usage Sensitive Three-way Calling
*72 - Call Forwarding Activation
*73 - Call Forwarding Deactivation
*74 - Speed Calling 8 - Change List
*75 - Speed Calling 30 - Change List
*76 - Advanced Call Waiting Deluxe
*77 - Anonymous Call Rejection Activation
*78 - Do Not Disturb Activation
*79 - Do Not Disturb Deactivation
*80 - Selective Call Rejection Deactivation
*81 - Distinctive Ringing/Call Waiting Deactivation
*82 - Line Blocking Deactivation
*83 - Selective Call Forwarding Deactivation
*84 - Selective Call Acceptance Deactivation
*85 - Calling Number Delivery Deactivation
*86 - Automatic Callback Deactivation
*87 - Anonymous Call Rejection Deactivation
*88 - Call Forwarding Busy Line/Don't Answer Deactivation
*89 - Automatic Recall Deactivation

*90 - Customer Programmable Call Forwarding Busy Line Activation
*91 - Customer Programmable Call Forwarding Busy Line Deactivation
*92 - Customer Programmable Call Forwarding Don't Answer Activation
*93 - Customer Programmable Call Forwarding Don't Answer Deactivation
*94 - Reserved For Local Assignment
*95 - Reserved For Local Assignment
*96 - Reserved For Local Assignment
*97 - Reserved For Local Assignment
*98 - Reserved For Local Assignment
*99 - Reserved For Local Assignment

NOTE: For rotary dial phones, dialing "11" can replace the use of the "*" on touch-tone phones.

Appendix III Acknowledgements

The following individuals are recognized for their contributions to this Technical Report (sorted by company name):

Mike Emmendorfer, Arris

Gordon Li, Broadcom (Lead Editor)

Pete Quigley, Cedar Point

Rob Lund, Cedar Point

Eric Turcotte, Ericsson (Lead Editor)

Mark Stewart, MetaSwitch

Johannes Schoepf, Nokia Siemens

Bill Gentry, Nortel Networks Cable Solutions, Inc.

Guy Vonderweidt, Nortel Networks Cable Solutions, Inc.

Mark Trayer, Samsung

Rajeev Seth, Sonus

Richard Wikoff, Sonus

Geoff Devine, Whaleback Systems

David Hancock, CableLabs

Sandeep Sharma and the PacketCable Architecture Team (CableLabs)