

**PacketCable™**

# **Residential SIP Telephony Feature Definition Technical Report**

**PKT-TR-RST-V03-071106**

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## Abstract

This technical report describes the PacketCable Residential SIP Telephony feature definitions and underlying assumptions. Residential SIP Telephony features include many of the features commonly available to residential subscribers of public switched telephone network services. Examples include caller ID, call waiting, speed dialing, and emergency calling capabilities. This document also describes the expectations for the devices that can be used for Residential SIP Telephony.

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# 1 INTRODUCTION

## 1.1 PacketCable Residential SIP Telephony Overview

The PacketCable Residential SIP Telephony effort is intended to replicate many of the residential features currently available on the public switched telephone network (PSTN), except that the features operate using a PacketCable-based IP (Internet Protocol) network using SIP (Session Initiation Protocol) for VoIP (voice over IP) sessions. The Residential SIP Telephony features described in this document are intended to form the basis for specification requirements that define the behavior of SIP clients that interact with the PacketCable network and the corresponding network behavior.

This technical report describes the behavior of residential telephony features using a standard RST Client (called RST User Equipment, or RST UE, or simply UE in this document) connected to the PacketCable network.

## 1.2 PacketCable Residential SIP Telephony Motivation

Prior PacketCable efforts have defined residential telephony capabilities that operate on a PacketCable IP network. PacketCable 1.5 defined an MGCP-based (Media Gateway Control Protocol) architecture in which a centralized Call Management Server (CMS) controls the telephony services provided by MTAs (Multimedia Terminal Adapters).

The PacketCable Residential SIP Telephony specification defines how UEs can support residential telephony capabilities similar to that of PacketCable 1.5. This allows for the deployment of new types of clients and services while maintaining continuity with existing residential telephony services.

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### 3 TERMS AND DEFINITIONS

PacketCable Specifications and Technical Reports use the following terms and definitions:

<b>Jitter</b>	Variability in the delay of a stream of incoming packets making up a flow such as a voice communication.
<b>Latency</b>	The time taken for a signal element to pass through a device or circuit.
<b>Media Gateway</b>	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.
<b>Off Hook</b>	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.
<b>On Hook</b>	The idle state of a traditional telephone, while no call is in progress and the telephone handset is sitting in its cradle.
<b>SIP Client</b>	The functional element used by subscribers to attach to the PacketCable network.
<b>Trunk</b>	An analog or digital connection from a circuit switch that carries user media content and may carry voice signaling (MF, R2, etc.).

## 4 ABBREVIATIONS AND ACRONYMS

PacketCable Specifications and Technical Reports use the following abbreviations and acronyms:

<b>3WC</b>	Three Way Calling
<b>AC</b>	Auto Callback or Alternating Current
<b>ACR</b>	Anonymous Call Rejection
<b>AR</b>	Auto Recall
<b>AS</b>	Application Server
<b>BCT</b>	Blind Call Transfer
<b>BLV</b>	Busy Line Verification
<b>CCT</b>	Consultative Call Transfer
<b>CFBL</b>	Call Forwarding Busy Line
<b>CFDA</b>	Call Forwarding Don't Answer
<b>CFV</b>	Call Forwarding Variable
<b>CID</b>	Caller Identity Delivery
<b>CIDB</b>	Caller Identity Delivery Blocking
<b>CIDS</b>	Caller Identity Delivery Suppression
<b>CNAM</b>	Calling Name
<b>CND</b>	Calling Number Delivery
<b>COT</b>	Customer Originated Trace
<b>CPE</b>	Customer Premise Equipment
<b>CSCF</b>	Call/Session Control Function
<b>CT</b>	Call Transfer
<b>DC</b>	Direct Current
<b>DN</b>	Directory Number
<b>DND</b>	Do Not Disturb
<b>DTMF</b>	Dual Tone Multi Frequency
<b>DVA</b>	Digital Voice Adapter
<b>E-DVA</b>	Embedded Digital Voice Adapter
<b>EI</b>	Emergency Interrupt
<b>E-MTA</b>	Embedded Multimedia Terminal Adapter
<b>FSK</b>	Frequency Shift Key
<b>IETF</b>	Internet Engineering Task Force
<b>IMS</b>	IP Multimedia Subsystem
<b>IP</b>	Internet Protocol
<b>ITU</b>	International Telecommunication Union
<b>LCD</b>	Liquid Crystal Display
<b>MGC</b>	Media Gateway Controller
<b>MTA</b>	Multimedia Terminal Adapter
<b>NAT</b>	Network Address Translation
<b>NCS</b>	Network Call Signaling

<b>PSTN</b>	Public Switched Telephone Network
<b>QoS</b>	Quality of Service
<b>RACF</b>	Remote Activation of Call Forwarding
<b>ROH</b>	Receiver-Off-Hook
<b>RST</b>	Residential SIP Telephony
<b>RTP</b>	Real-time Transport Protocol
<b>SCA</b>	Selective Call Acceptance
<b>SCF</b>	Selective Call Forwarding
<b>SCR</b>	Selective Call Rejection
<b>SIP</b>	Session Initiation Protocol
<b>SLE</b>	Screening List Editing
<b>SPP</b>	Subscriber Programmable PIN
<b>TDD</b>	Telecommunications Device for the Deaf
<b>UE</b>	User Equipment
<b>URI</b>	Uniform Resource Identifier
<b>URL</b>	Uniform Resource Locator
<b>VMWI</b>	Visual Message Waiting Indicator
<b>VoIP</b>	Voice over Internet Protocol

## 5 PACKETCABLE RESIDENTIAL SIP TELEPHONY SCOPE

The PacketCable Residential SIP Telephony Feature Definition Technical Report describes a grouping of telephony call features and capabilities that approximate the services available to most PSTN residential voice subscribers. Residential SIP Telephony 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 IMS) leverages the use of triggers within the core network to determine when application level processing is required.

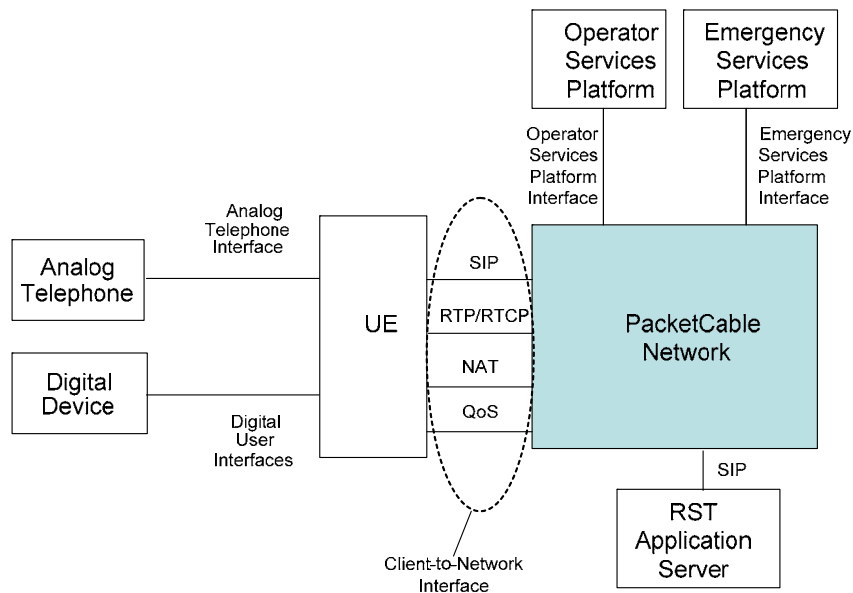
Common groupings of features have been proposed in this document so that implementers can easily refer to groups of related features and capabilities when describing their implementations.

This section provides:

- An overview of the PacketCable Residential SIP Telephony feature capabilities.
- A description of the main feature groupings.

### 5.1 Overview

This document defines Residential SIP Telephony 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 for this document, and the interfaces that are within scope for this document.



**Figure 1 - Residential SIP Telephony Feature Definition Context**

This document defines requirements for the Residential SIP Telephony capability set as they impact the following elements:

**UE:** the functional element subscribers use to attach 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 destruction of sessions.
- Media: For carrying audio or other media traffic, and related statistics.
- Network Address Translation (NAT) Firewall Traversal: For permitting Residential SIP Telephony features to operate through NAT Firewall devices.
- Security: For securing interfaces.

**Analog Telephone:** A widely deployed residential telephone device. This document describes Residential SIP Telephony requirements on the interface between the UE and the Analog Telephone (the Analog Telephone Interface), for those devices that support a traditional Analog Telephone interface.

**Digital Device:** This Feature Definitions document does not specify requirements for implementations of user interfaces for digital devices (devices other than traditional analog telephones). Digital devices may interact with the UE according to the service provider or vendor definitions.

**RST Application Server:** The Residential SIP Telephony Application Server is a logical network component that implements some aspects of RST features. The PacketCable architecture provides a framework for triggering SIP messages to the application server based on the service profile of a user.

**Operator Services Platform:** Some Residential SIP Telephony features require the intervention of an operator. This document describes the interactions between the PacketCable network and the Operator Service Platform.

**Emergency Services Platform:** Residential SIP Telephony features require specialized capabilities supporting emergency calling. This document describes the interactions between the PacketCable network and the Emergency Services Platform.

## 5.2 Residential SIP Telephony Feature Groupings

The Residential SIP Telephony 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, and three-way calling.
- Miscellaneous Features: including distinctive alerting, 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.

## 6 PACKETCABLE RESIDENTIAL SIP TELEPHONY FEATURE DEFINITIONS

This section contains detail on each of the call features that may be offered as part of PacketCable Residential SIP Telephony. Service definitions are described for the simple case of a single UE emulating the legacy PSTN residential voice service, and for the case of how similar goals can be met in a PacketCable SIP-enabled environment, where multiple UEs can be a part of the same Residential SIP Telephony subscription.

The Residential SIP Telephony suite of specifications is embodied in a separate standalone release. The suite consists of the following specifications:

- PacketCable Residential SIP Telephony Feature Specification

This specification defines specific requirements on the UE and Application Servers to implement Residential SIP Telephony 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 Residential SIP Telephony features are specified in [CMSS1.5].

- PacketCable Residential SIP Telephony Accounting Specification

The purpose of this specification is to define the collection of usage data needed to support Accounting of Residential SIP Telephony Features.

- PacketCable Residential SIP Telephony E-DVA Specification

This specification defines the embedded Digital Voice Adaptor (E-DVA) requirements for the analog interface and for powering of the E-DVA. A DVA is a UE that supports an analog phone interface. An embedded DVA is a DVA integrated with a cable modem.

### 6.1 Basic Calling Capabilities

#### 6.1.1 UEs and Public Identities

One of the major differences between a traditional telephone network and a SIP Telephony network lies in the relationships between users, public identities, UEs, and devices. A user may have multiple devices, each of which may host multiple UEs, and each UE is registered to one or more public identities. A public 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 identity is generally associated with a user.

In traditional residential telephone services, a telephone number (in SIP terms, a public identity) has traditionally been associated with one or more telephone lines to a user's home, and in that user's home one or more telephones may be connected to the same telephone line. When a call is placed to the telephone number (public identity), all of the telephones connected to the telephone line ring, and any of the phones may be picked up to listen or speak on the same call (the phones act as "bridged line appearances").

In SIP-based telephone networks, a single telephone number (public 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 in the household 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 identity to ring, but the first one to pick up gets the call, and the other UEs do not necessarily have the capability to listen and speak in the same telephone conversation.

### 6.1.2 Basic Call for Multiple UEs per Public Identity

This section describes the behavior of a basic call in the case where there are two or more UEs behind a single public identity. There are at least three distinct categories for UE behavior when multiple UEs are behind a single public identity.

- Multiple Call Arrangement (MCA) – allows the UEs to operate independently from one another, but sharing the single public identity.
- Single Call Arrangement (SCA) – allows multiple UEs to handle only one call for the public identity at any given time.
- Extension Bridging (EXB) – UEs are automatically bridged together if a second UE answers when an active stable call already exists for the public identity.

**Note:** The first phase of RST specifications assumes that there is a single UE behind a given public identity.

### 6.1.3 Addressing

#### 6.1.3.1 Dial String Entry

When a legacy PSTN telephone device is used in combination with an UE, it is customary for a call to be initiated by entering a dial string that denotes the target of the call and for features to be invoked by entering a dial string that denotes the feature requested as well as data pertinent to that feature. Normally the end of the dial string is determined when a fully formed syntactic unit designating a call target or feature has been entered (or after a digit-entry timer has expired), without the need for an end-of-dialing indicator. However mobile phones have established an alternate precedent where the end of the dial string is signaled by pressing a special button. UEs are generally expected to use either syntax-matching (with digit entry timers), or may use an end-of dialing button instead to determine when a complete dial string has been entered by a user.

Dial strings are strings of characters that correspond to DTMF digits. (These are the characters 0-9, A-D, \*, #.) A syntax applied to dial strings determines which dial strings denote call targets and which denote features. Strings which denote neither call targets nor features are invalid. UEs recognize incorrect dial strings as soon as an invalid digit is entered (based on the provisioned digit map) and provide immediate feedback to the user.

Service providers can provision UEs with configuration data that allows the UE to analyze the correct syntax for dial strings, and configures the UE's timers for initial digit entry, maximum inter-digit timing, short inter-digit timing and digit detection timing.

#### 6.1.3.2 Initiating a Call or Feature via Dial String

With traditional analog telephone devices it is customary for features to be invoked by entering a dial string that denotes the feature requested as well as data pertinent to that feature. Features are often, but not always, denoted by star-codes (dial strings that begin with "\*", see Appendix II) whereas dial strings consisting exclusively of digits usually denote a request for call initiation. In PacketCable calls are initiated by a UE, while features may be performed by the UE, an element in the network, or a combination of the two.

From the perspective of the UE, it may be necessary to distinguish dial strings that denote call targets from those that denote features to be implemented by a network element. Either may require a similar action on the part of the UE – encode the dial string as a target address and initiate a call to that address. For some feature dial strings, though, the UE may be expected to perform a special UE action, and not create a network session. Service providers can provision UEs with digit maps that allows the UE to determine which dial strings should be processed locally in the UE, and which feature dial strings should be delivered to the network for processing.

### 6.1.3.3 *Creating a Routable Address from a Dial String*

PacketCable Residential SIP Telephony is intended to serve as an alternative to conventional PSTN service. This requires that it support the same kinds of dial strings used in the PSTN, and that it support calling addresses in the PSTN.

The dial strings conventionally used in the PSTN are often context sensitive and non-unique:

- A single dial string may denote different targets, depending on the source of the dial string, (e.g., 7-digit dialing).
- Several different dial strings may denote the same target, (e.g., 555-1234, 978-555-1234, 1-978-555-1234, 011-1-978-1234).

To successfully route requests it is essential for a UE to remove ambiguity in the dial string that was entered by the user. The syntax patterns and rules used to interpret the dial string entered by the user and for the corresponding client action will be part of the provisioned digit map.

### 6.1.4 **Call Progress Indications**

Call progress indications are signals that inform the user of call progression status. Call progress indications are applied by the originating UE to inform the user of the status of an attempted call, usually in the form of tones played to the calling user through the earpiece of the telephone handset (in the case of a traditional telephone user interface). Call progress indications can also be used to notify a user that a feature is activated or deactivated. An example would be the transmission of a stutter dial tone.

For UEs that support a traditional analog telephone port interface, the standard call progress tones that emulate the North American PSTN network are defined in [GR 506]. Referenced in the industry as the Precise Tone Plan, these tones are a continuous or cadenced tone or tone pair using the frequencies 350, 440, 480, and 620 hertz. UEs supporting analog (RJ11) access lines that emulate the PSTN network are expected to locally generate call progress tones, including ring back, busy, and all of the standard error tones defined in [GR 506]. Call progress tones defined for PacketCable are listed in Appendix I.

### 6.1.5 **Announcements**

In the course of traditional processing of PSTN telephone calls, voice announcements have been played to the user to indicate error conditions or to prompt the user to enter additional information. In traditional telephone networks, these announcements were stored in the network and played to the user from network equipment when needed. An example of a voice announcement played to callers for is Solicitor Call Blocking, where callers are instructed to hang up if they are a solicitor, or press a key to claim they are not a solicitor.

Some UEs have the capacity to store and play announcements (tones, voice or music) on the local device. While most announcements for most of the features described in this document are expected to be network-resident, service providers may elect to have the UE store or generate announcements locally. Service providers can provision announcements to be locally-resident on UEs, and can provision UEs to use these locally-resident announcements when appropriate. For example, if local announcements are supported on the device, a locally generated announcement may be used to notify the user when the network is unavailable (see Section 6.12.1 for conditions where the network would be considered in-service or out-of-service).

Regardless of whether an announcement is stored/generated on the UE or in the network, each public identity can be provisioned with a preferred language selection for the announcements they hear. The language selection is associated with the public identity of the user that subscribed to Residential SIP Telephony, and, where possible, the language selection is used to avoid prompting the user for their preferred language in situations where their language can already be known from the user's configuration data.

### 6.1.6 Permanent Sequence and Lockout

Permanent Sequence and Lockout describe the response required if an access device is left in a state where it is requesting service for an extended period of time. In the case of an analog telephone client, for example, the telephone handset is left off-hook for an extended period of time. This section describes the behavior of an analog telephone client, with the understanding that other implementations may make more sense for other client types.

There are generally two ways a handset can be left off-hook:

- **Origination Mode:** the device was in an idle state (on-hook) and is placed off-hook and left there (presumably accidentally).
- **Termination Mode:** the device was in use for a call and was never successfully returned to an idle state (placed on hook). This can occur whether the party was the originator or the receiver of the original call.

In the origination mode, the UE initially offers dial tone to the end user to prompt the end user to dial the desired number. After a provisionable timeout period expires, before which the user could have committed a dialing error or simply did not dial at all, the UE invokes its permanent sequence and lockout treatment.

In the termination mode, the UE applies an audible error signal (for example, in the ear interface of a handset) for a provisionable time period or until an on-hook state is detected. If the provisionable time period expires before the on-hook state is detected, the permanent sequence and lockout treatment is invoked by the UE.

When the permanent sequence and lockout treatment is invoked, the UE begins to generate a configurable sequence of tones and announcements, which may include the open-loop signal to shake off modems and answering machines. For example, the "Receiver-Off-Hook" (ROH) tone (sometimes called Howler Tone) has traditionally been used in this situation. This is a four frequency tone sequence which increases in amplitude and repeats for a provisioned period of time. This tone can be re-defined by provisioning a different tone to the UE.

The sequence of tones and announcements is generated until a timer expires, after which the sequence stops and no analog line operation is accepted (Lockout) until the phone is placed on hook for a provisionable time period (a "reset"). After this "reset", the analog line returns to normal operation, responding to the next originator mode off hook state with dial tone or in termination mode with appropriate (ring, caller ID, etc.) signaling.

### 6.1.7 No Answer Timeout

The "No Answer Timer" prevents network resources from being tied up waiting for an answer event that is not likely to happen. The called party is given a reasonable amount of time to answer the call and if the answer does not occur within the "No Answer Timer" time period, the call attempt is ended. The "No Answer Timer" is configurable on the UE.

The "No Answer Timer" is started when the terminating UE receives a request to establish a session. The timer is cancelled if the terminating UE answers prior to expiration of the "No Answer Timer". If the "No Answer Timer" expires before the terminating UE answers, alerting is discontinued at the terminating UE and the originating UE receives an indication that the call cannot be completed. The terminating UE cannot answer the call once the "No Answer Timer" has expired.

User treatment on no-answer timeout is a provisionable parameter. Possible user treatments include an error tone or a voice announcement, played by the calling party's network or by the calling party's UE when the no-answer error indication is received.

The "No Answer Timer" may need to be set with awareness of how it may impact features such as call forwarding don't answer (CFDA).

### 6.1.8 UE Status

When a UE knows it has become available or unavailable to receive calls, it signals its availability or unavailability to the PacketCable network by registering (or de-registering) with a configured public identity.

If a call is placed to a public identity for which there are no registered UEs, then the public 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 for features that are defined in terms of "busy" or "no answer" conditions, such as Call Forwarding Busy Line (CFBL), CFDA, and call forwarding to voice mail. In the case of the CFBL feature, an unreachable UE does not trigger CFBL forwarding, but may trigger call forwarding to voice mail or CFDA.

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

### 6.1.9 Quality of Service

PacketCable Residential SIP Telephony utilizes the Quality of Service (QoS) capabilities of the PacketCable architecture as defined in [ARCH-FRM TR]. In general, if QoS resources can't be allocated for a call, operator policy dictates whether the call is permitted to proceed or not. One operator may not permit calls for which QoS cannot be allocated to proceed while another operator may permit such calls. In the case of an emergency services call, or an operator services emergency interruption, the call may be attempted even if QoS resources cannot be allocated for the call, per operator policy. Also, per operator policy, emergency calls may be granted higher-priority QoS than other, non-emergency calls.

## 6.2 Caller ID

Caller ID features govern the delivery (or blocking) of the caller's number and name to the called party. The called party may permit (or disallow) the display of the caller's number and name, and the calling party may prohibit (or permit) the delivery of his number and name to the called party. In the case of Anonymous Call Rejection (ACR), the called party can automatically reject calls from parties that do not supply their identity to the called party.

In SIP Telephony networks, the caller's identity may not be a telephone number, and could well be an alphanumeric address in the form of a Universal Resource Identifier (URI). Even if the caller's identifier is non-numeric, the called UE displays the identifier that is signaled to it.

### 6.2.1 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.

For UEs with a traditional analog telephone interface, there are two ways to deliver the CID (CND and CNAM):

- On-hook CID delivery with power ringing – When the called subscriber line is on-hook, the calling party information is delivered during the long silent interval of the first ringing cycle through FSK.
- Off-hook CID delivery – When the called subscriber is off-hook, the calling party information is delivered using FSK.

[GR 30] specifies the generic requirements for transmitting asynchronous voice band data to CPE.

The CNAM and CND features are implemented (or disallowed) by the UE according to provisioning by the service provider.

Calling Number Delivery treatment for UEs that provide other interfaces, such as SIP phones with LCD displays and soft clients, are not defined.

### **6.2.2 Caller ID Per-Line Blocking**

Calling Identity Delivery Blocking (CIDB) allows subscribers to control whether or not their calling number and calling name information is delivered with outgoing calls by setting the value of the Permanent Presentation Status for calling number and calling name delivery. The feature can be invoked on a per-line basis.

The CIDB feature affects the Presentation Status of the caller's name and number identity information. The Presentation Status is a flag that indicates whether it is permissible to deliver the information to the called party. There are two types of Presentation Status flags. Permanent Presentation Status (PPS) is a stored attribute about the subscriber line. Per-Call Presentation Status (PCPS) has significance only to the current outgoing call. It is generated as a function of the Permanent Presentation Status and CIDB feature invoked for the call (if any). The presentation status is associated with both the subscriber line's Calling Number (directory number) and Calling Name.

At subscription time, the subscriber requests the service provider to provision default values for the Permanent Presentation Status flags. The possible values are 'PUBLIC' and 'ANONYMOUS'. A value of 'PUBLIC' indicates that the Caller ID information (Caller Name and Caller Number) should be delivered to the call terminator as the default. A value of 'ANONYMOUS' indicates that the Calling Identity information should not be delivered to the call terminator as the default. All subscribers have the option to request values for the Permanent Presentation Status flags.

### **6.2.3 Caller ID Per-Call Blocking**

On a per-call basis, a subscriber with the Caller ID Suppression (CIDS) feature (as defined in [GR 391]) enabled can override the default values for the Permanent Presentation Status flags, thereby forcing delivery of either the Caller ID information (both Caller Name and Caller Number) using the \*82 feature activation code, or to block it using the \*67 feature activation code.

### **6.2.4 Caller ID with Call Waiting**

The Caller ID with Call Waiting feature is defined for the case where there is already a call in progress to a UE, and a second call is placed to the same UE. If the called party's UE has the Call Waiting (CW) feature enabled, it not only plays the call waiting tone to indicate the incoming call; it also displays the identity of the calling party (if the calling party's identity is provided). The off-hook CID procedure defined in Section 6.2.1 is used to communicate the identity of the caller.

### **6.2.5 Anonymous Call Rejection (ACR)**

Anonymous Call Rejection (ACR) is the ability for a called party's UE to automatically reject incoming calls that do not supply the identity of the calling party (when, for example, the calling party has subscribed to the Caller ID Per-Line Blocking feature). ACR is considered as an optional feature, and as such can be enabled for a specific user based on prior arrangement between the PacketCable network provider and the subscriber. ACR may be disabled for administrative reasons.

When the called party's ACR feature is activated, the called party UE should not be alerted when the calling party has not provided an identity. The calling user is given an appropriate indication that the called user has rejected the call due to the application of ACR.

The user can enable (activate) and disable (deactivate) ACR feature, as well as interrogate the status of ACR service.

Some authorized callers may not be subject to ACR. For example, an emergency operator that anonymously calls back a user that has enabled the ACR service should not be blocked. This requires a mechanism to override the ACR service if the caller is an authorized emergency operator.

The ACR feature is described in [GR 567].

## 6.3 Call Forwarding

Call Forwarding features divert a call from its original target to a new, revised target, based upon rules associated with the called party. The conditions for forwarding a call can be situational, as in the case of Call Forwarding on Busy Line (CFBL) or Call Forwarding on Don't Answer (CFDA), or the call forwarding can be unconditional, as in the case of Call Forwarding Variable (CFV).

There are several different mechanisms for activation of call forwarding that are described in [GR 586]. The call forward programming/activation mechanisms in force for CFV, CFDA and CFBL can be provisioned by the service provider.

Call forwarding can be setup to any address that a subscriber can successfully call. Calls cannot be forwarded to addresses that can only be dialed with blocking overrides, because blocking overrides are meant to be invoked by the calling party, and in the case of call forwarding the party that subscribed to the blocking feature is not on the call and not available to enter the correct blocking override code.

All call forwarding features operate on a public identity basis. This means that forwarding programmed from any UE behind a public identity affects all UEs behind that public identity.

The calling number or name presented after a call is forwarded by any of the forwarding features is always that associated with the original caller, not the forwarding party.

For each of the Call Forwarding features, there is a danger of forwarding loops, where a call is forwarded in circles around the network indefinitely, posing the danger of over-allocation of network resources and an increased burden of signaling traffic. Networks should provide mechanisms for the detection of forwarding loops to prevent the damaging effects of such behavior.

The analog user interfaces for each of the Call Forwarding features are described in [GR 580], [GR 586], [GR 220], and [GR 217]. UEs with other interfaces are allowed to provide alternative user interfaces because different types of clients have widely varying user interface capabilities. In addition to the wide range of clients, a service provider may offer a web client mechanism to allow a subscriber to manage his or her feature data. For this section, however, descriptions are limited to interactions with the analog telephony client, rather than a feature management web client or other client type.

### 6.3.1 Call Forwarding Variable (CFV)

Call forwarding variable (CFV) is a feature that allows a subscriber to forward all calls to the subscriber's public identity to another location. With CFV the forwarding happens immediately and the forwarded public identity has no opportunity to answer the call prior to the forwarding. The analog user interface for the CFV feature is described in [GR 580].

Call forwarding can be broken down into several distinct operations: programming the forwarding number and activating forwarding, deactivating forwarding, and the actual forwarding of incoming calls.

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 UE.

When call forwarding variable is active, all calls to the public identity are immediately forwarded to the programmed forward-to number. Each time a call is forwarded, the forwarded UE provides an indication to the user (e.g., ring splash) that a call was just forwarded. This serves as a reminder to the subscriber that they have forwarding activated. The CFV feature works even when the called party has no UEs reachable on the network.

The feature codes (star codes) used for activating, programming, and deactivating the CFV feature can be provisioned by the service provider.

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

### **6.3.2 Call Forwarding Don't Answer (CFDA)**

Call forwarding don't answer (CFDA) is a feature that allows a subscriber to forward calls to the subscriber's public identity to another location if the calls are not answered within a given number of rings or time period. If the subscriber answers the incoming call prior to the specified number of rings or expiry of the time period, the subscriber is connected to the incoming call. If the subscriber does not answer the incoming call within the specified number of rings or expiry of the time period, the call is forwarded to the forward-to number. Until the call is answered (either by the originally called UE or by a forward-to UE), the caller hears audible ringing. The analog user interface CFDA feature is described in [GR 586].

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 UE.

When call forwarding don't answer is active, all calls to the public identity are forwarded to the programmed forward-to number if the call is not answered at the public identity prior to a specified number of ring cycles or expiry of defined time period.

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

The CFDA feature works even when the called party has no UEs on the network.

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

### **6.3.3 Call Forwarding on Busy Line (CFBL)**

Call forwarding on busy line (CFBL) is a feature that allows a subscriber to forward calls to the subscriber's public identity to another location if the public identity is busy. With CFBL the forwarding happens immediately and the forwarded public identity has no opportunity to answer the call prior to the forwarding. The analog user interface for CFBL feature is defined in [GR 586].

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 UE.

When call forwarding busy is active, calls to the public identity, when the public identity is busy, are immediately forwarded to the programmed forward-to number.

If there are multiple UEs behind a public identity and not all UEs are busy, then the forwarding waits until any ringing UEs have a chance to answer the call.

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

When the called subscriber has no registered UEs on the network, the public identity is considered "unreachable" rather than "busy", so CFBL is not invoked. When the target UE is in lockout state, the UE is considered "busy", so if CFBL is active for the public identity (and if the UE in lockout state is the only UE registered for this public identity), then CFBL is invoked.

#### **6.3.4 Selective Call Forwarding (SCF)**

Selective call forwarding (SCF) is a feature that allows a subscriber to forward selected calls to the subscriber's public identity to another location. With SCF the forwarding happens immediately if the calling line identification matches a number in the SCF screening list. If the calling line identification is not available or is not in the SCF screening list, the incoming call is given normal termination treatment and may be subject to other features. If a call is forwarded by SCF, the forwarded public identity has no opportunity to answer the call prior to the forwarding. The analog user interface for the SCF feature is described in [GR 217].

An SCF subscriber can use the Subscriber-Programmable PIN (SPP) feature to program a PIN code for SCF. The SPP feature is described in [GR 220].

The SCF feature works even when the called party has no UEs on the network.

#### **6.3.5 Remote Activation of Call Forwarding**

The remote activation of call forwarding feature (RACF) allows a subscriber who also subscribes to call forwarding variable to control CFV for one of his UEs from another location (i.e., not at the UE being forwarded). In order to prevent unauthorized forwarding, the subscriber is required to provide a PIN or password when activating or deactivating RACF. PIN numbers can be specified by the service provider. The service provider may also allow the subscriber to create or modify his PIN numbers using the SPP feature.

To activate RACF, the subscriber places a call to the RACF service access address. The subscriber is then prompted to enter the "N" digit telephone number to be forwarded (the number of digits and form of digits to enter may be set by the service provider, and should be indicated in the prompts), followed by their PIN number. If the subscriber enters a telephone number and corresponding PIN, they are then prompted for the CFV activation or deactivation code. If the subscriber's telephone number and PIN do not match, then the feature may allow a number of retries so that the subscriber can attempt the number/PIN combination again. If after the maximum number of retries, the subscriber still has not entered a valid number/PIN combination, the subscriber is informed that RACF programming has failed and the call is terminated.

If after entering a valid number/PIN combination, the subscriber enters the CFV deactivation code, CFV is deactivated and the subscriber is given a confirmation indication (normally confirmation tone) and the call is ended.

If after entering a valid number/PIN combination, the subscriber enters the CFV activation code, the response depends on the type of CFV that the subscriber has. If the subscriber has fixed CFV (non-programmable), the subscriber hears confirmation tone and the call ends. If the subscriber has programmable CFV the subscriber is prompted for the forward-to number.

Once the subscriber enters a forward-to number, the number is read back to the subscriber for verification. The subscriber then has the option to confirm the forward-to number or enter another number. If the subscriber confirms the forward-to number, they receive confirmation indication and the call is ended. If the subscriber requests to enter a different number, they are prompted for the forward-to number digits again.

If the RACF programming fails for any reason, the CFV status is not to be changed.

If a subscriber successfully deactivates RACF, but CFV was not activated in the first place, the subscriber gets confirmation indication anyway.

If a subscriber already has CFV activated when they successfully program RACF, the RACF programming overrides the CFV programming and the subscriber gets confirmation indication.

If a subscriber dials an invalid forward-to number while programming RACF (as described in the CFV section), the subscriber is prompted to enter another number.

If the subscriber disconnects prior to hearing confirmation tone on an RACF programming attempt, the CFV status remains unchanged.

RACF can be programmed from anywhere that can call the RACF service access address (or anywhere that allows access to the service provider's web programming page).

All RACF announcements should be interruptible so that subscribers familiar with the prompts are not forced to wait for announcement completion.

If any of the following conditions occur during an RACF programming attempt, the caller gets a failure indication and the call is terminated:

- Subscriber fails to enter a valid number/PIN combination in the allowed number of attempts.
- Subscriber fails to enter a valid CFV activation or deactivation code when prompted.
- Subscriber fails to enter a valid CFV forward-to number during RACF activation in the allowed number of attempts.
- Subscriber fails to enter a valid forward-to number verification choice in the allowed number of attempts.
- Caller is not subscribed to RACF.
- Caller is not subscribed to CFV.

The subscriber PIN may be provisioned by the operator or set by subscriber action, or both, depending on service provider policy. Service providers are able to view and reset passwords.

Subscribers may enter the forward-to number using the speed dial feature, if available to the subscriber.

### **6.3.6 Call Forwarding to Voice Mail**

Call forwarding to voice mail uses the forwarding mechanisms already described (CFV, CFBL, CFDA, or SCF).

Voice mail systems need to distinguish between calls made directly to them versus calls forwarded to them. If a call is forwarded to a voice mail system, the voice mail system needs to determine the address that forwarded the call for use in determining the proper mailbox. Also, the voice mail system needs to determine if the call was forwarded via CFDA or CFBL. If a voice-mail system is called directly, it needs to determine the calling address for use in identifying the caller.

## **6.4 Call Blocking**

Call blocking features provide usage limitations (and hence billing limitations) on call originations and call terminations. For PacketCable the option exists of applying these features to all UEs registered to the public identity being billed, or to the individual UEs on a SIP device URI basis.

### 6.4.1 Outbound Call Blocking

The Outbound Call Blocking feature blocks specified categories of calls from being placed from a UE. It is activated through subscriber provisioning.

The following types of destination limitations can be applied:

- Direct-dialed international calls (011+)
- Direct-dialed calls (1+)
- Operated-assisted calls (0+)
- Operator-assisted international calls (01+)
- Operator-request calls (0-, i.e., 0 followed by a # or 4-second timeout)
- Local directory assistance calls (411)
- Long distance directory-assistance calls (555)
- Carrier access code (+10XXX)

This feature may bundle all of the above or selected individual restrictions. For example, long-distance calls (1+) may be allowed, but international calls (011+) may be blocked. A PIN may be provided to allow a subscriber to remove the restrictions on a per-call basis. If the PIN is authenticated, the call proceeds as a normal call; if not, the originating caller is connected to a media server for an intercept announcement or simply provided with re-order tone.

When an outbound call is blocked, the calling party either hears an error tone or an announcement.

[GR 564] describes the Code Restriction and Diversion feature upon which this PacketCable Residential SIP Telephony feature is based.

### 6.4.2 Collect Call Blocking

This feature prevents the subscriber from receiving collect calls. This call termination blocking feature prevents charges for calls not originating from the subscriber's telephone number.

### 6.4.3 Solicitor Blocking

The solicitor blocking feature is a type of incoming call screening designed to reduce the number of solicitor calls a subscriber receives. Two variations of this feature are considered.

The first option is one in which anonymous callers are presented with an announcement indicating that the called party does not accept solicitor calls and to add the number to the do-not-call list, or press one or wait to attempt to connect to the called party.

The second option is to ask anonymous callers to record their names, which can then be played to the called party. If the name is recorded, the called party is presented the call and given the option to accept, ignore, or direct the call to a solicitor blocking announcement, a generic blocking announcement, or to voice-mail. If the anonymous calling party does not record a name, then the called number is not alerted. The screening options are left up to the service provider to define.

Incoming calls that are not anonymous bypass the Solicitor Blocking feature and terminate to the subscriber.

This feature has no mandatory subscriber interface for activation or deactivation. Once the subscriber orders the feature from the service provider the feature is activated and remains so until the subscriber explicitly cancels the feature.

## 6.5 Multi-Party Features

### 6.5.1 Call Waiting

For convenience, the following functional description of the Call Waiting feature is given with respect to UEs that support the analog phone interface. However, the description also applies to other types of UEs such as soft SIP clients and WiFi SIP phones, etc., with the tones and buttons specific to the analog phone being replaced by the counterpart user-interface elements specific to a particular UE type. Except where noted, this feature conforms to [GR 571] and [GR 572].

The Call Waiting feature is initially activated when the service provider provisions the feature in the UE.

Call Waiting occurs during an active call. When one of the parties in the active call has subscribed to the Call Waiting feature, and a second call is offered to the party, then the party hears a call waiting tone to alert them to new offered call. If the called party has subscribed to the Call Waiting with Caller ID feature, then the caller's ID is displayed for the called party. The called party may choose to ignore the second incoming call, or may answer the call and place the pre-existing call on hold. The called party may then switch back and forth between the first and second callers.

When the Call Waiting tone alerts the user of the UE that there is a pending incoming call, the user can take one of the following actions:

1. Disconnect from the Current Call and Answer the Waiting Call. The user ends the current call by going on-hook. The client then rings to indicate the waiting call. The user answers the waiting call by going off-hook (as for a normal incoming call). Before the waiting call is answered, the remote caller continues to hear the normal ringing tone.
2. Place the Current Call on Hold and Answer the Waiting Call. Until the caller has hung up or has been diverted, the called party can answer the waiting call by pressing the Flash Button temporarily and then releasing the button. The current call is then put on hold and the user is connected to the waiting call. Before the waiting call is answered, the remote caller continues to hear the normal ringing tone. The user can switch between the two calls by toggling the Flash Button.
3. Continue with the Current Call and Ignore the Waiting Call. The user can ignore the waiting call and continue with the current call. The remote caller continues to hear the ringing tone and is treated according to other features currently in effect for UE.

When the phone is off-hook without the call being established yet, any incoming call receives a busy tone and Call Waiting is not activated, unless there are other UEs that share the public identity with the current UE.

The Call Waiting feature can be deactivated in the following ways:

1. Turn off the Call Waiting Feature before Initiating a Call. Upon going off-hook for initiating a call, the user dials a star code (\*70 is the recommended star code). The UE then generates a short confirmation tone followed by the dial tone. The Call Waiting feature is then deactivated, and the user can proceed with the call initiation. The feature is automatically activated after the user goes on-hook.
2. Turn off the Call Waiting Feature during an Active Call. During a basic active call (when no call is waiting), the user presses the Flash Button, and then dials a star code (\*70 is recommended). The Call Waiting feature is deactivated for the current call, and the user then hears a short confirmation tone and is returned to the current call. The Call Waiting feature is activated after the user goes on-hook.

With the Call Waiting feature deactivated, the remote caller of the second call is provided with a treatment consistent with the currently active features, and the local UE does not generate the Call Waiting tone.

The number of simultaneous instances of Call Waiting that are permitted is a provisionable data element per UE. UEs may also have a maximum upper bound for the number of supported simultaneous instances of Call Waiting.

When an incoming call is put on hold by one UE, the call can not be picked up by other UEs registered to the same public identity.

When one UE is off-hook without the call being established yet and a call comes in, an incoming call should alert other UEs registered to the same public identity.

When multiple UEs are supported per public identity and one of the UEs is engaged in an active call, another incoming call alerts all other UEs associated with the common public identity and the currently active UE applies the call waiting tone to its user.

### **6.5.2 Hold**

During an active call, the Call Hold feature allows the user to temporarily halt the playback of media between the called party and the calling party, without ending the call, for a provisionable time period. The call is kept active while the user may switch to perform other tasks.

Call Hold is activated and deactivated via UE-specific mechanisms. For instance, for a UE supporting the analog phone interface, the feature may be activated and deactivated by toggling the Flash Button on the phone, receiving recall dial tone, dialing the hold star code (\*52 is one example that has been used), receiving another recall dial tone, and having the option of dialing another party or activating another feature, as described in Telcordia [GR 579]. To take a call off hold, the user toggles the flash button, receives recall dial tone, and dials the hold star code, and the held party again exchanges media with the user's UE. If a user has placed a call on hold and then hangs up on the call, the phone rings and when answered the user is able to speak to the party that was on hold. The hold feature does not specify music-on-hold as an option; although UEs may send some indication that the call is on hold to the held user (periodic tones, comfort noise, or music, as examples). Telcordia [GR 579] describes the analog user interface for the hold feature in more detail.

For analog phone clients, only one call may be held at a time.

For UEs that do not have an analog phone interface, the specific user interface requirements of Telcordia [GR 579] may be replaced with user interactions that provide similar results. For UEs that do not have an analog phone clients, user interfaces may be designed that allow multiple calls to be held.

Hold is a per-UE feature, and not a per-public-identity feature. For users who have multiple UEs registered using the same public identity, where one of those UEs has placed on hold, the call may not be picked up on another UE by dialing the hold star code on the other UE.

### **6.5.3 Call Transfer (CT)**

Call Transfer (CT) allows one party to transfer a second party (a participant in an active call with the transferring party) to a third party (a participant in another active call with the transferring party). The transferring party can choose to attend the transferred call (an attendant call transfer) or can elect to drop out of the transferred call (a blind call transfer).

The analog user interface for the call transfer feature is defined in Telcordia [GR 579], though only the following flavors of call transfer are intended for implementation as part of PacketCable Residential SIP Telephony:

- Blind Call Transfer (BCT)
- Consultative Call Transfer (CCT)

#### **6.5.3.1 Blind Call Transfer (BCT)**

The Blind Call Transfer (BCT) feature allows either party in a two-way call to put the call on hold and make another call. While the third party's phone is ringing, the party that makes the second call may join the second call with the party on hold without talking to the third party, and then hang up.

The call scenario of Blind Call Transfer (BCT) based on an analog phone is described as follows:

- UE A and UE B are on an active call.
- Subscriber B needs to transfer the call to UE C. If there is a CT button, subscriber B presses it. Otherwise, subscriber B can flash the hook.
- Subscriber B hears the stutter tone.
- UE B plays the dial tones.
- Subscriber B dials the phone number of subscriber C.
- Subscriber B's CID is delivered to subscriber C.
- There are several possibilities that can happen at this point:
  - Subscriber C's UE(s) are ringing. Subscriber B hears the ring back tone. Subscriber B needs to hang up to complete the transfer. If C's phone is still ringing, subscriber A can hear the ringing after subscriber B hangs up. If Subscriber B flashes before hanging up the phone, the session becomes a 3-way call. Subscriber B may then hang up the phone to complete the CT. When subscriber C answers the phone, subscriber A and subscriber C are able to talk to each other. From this point on, it is a normal two-way call between subscriber A and subscriber C.
  - If subscriber C's phone is ringing, but subscriber C does not answer the call, the scenario is described below.
  - If subscriber C's phone is busy, subscriber B can flash the hook to get back to subscriber A. In that case, the connection to subscriber C is terminated.

This feature is described in detail in [GR 579]. Please consult [GR 579] for details about call termination and call failure scenarios.

### **6.5.3.2 Consultative Call Transfer (CCT)**

The Consultative Call Transfer (CCT) allows either party in a two-way call to put the call on hold and make another call. After the third party (or attendant), answers the call, the party makes the second call talks to the third party and then join the call with the party on hold, and then may choose to hang up. After the attendant answers the call, the attendant becomes the controller.

The call scenario of Attendant Call Transfer (CCT) based on an analog phone is described as follows:

- UE A and UE B are on an active call.
- Subscriber B needs to transfer the call to subscriber C. Subscriber B flashes the hook to put UE A on hold.
- Subscriber B hears stutter tone.
- UE B plays the dial tones.
- Subscriber B dials the phone number of subscriber C, the attendant.
- Subscriber B's CID is delivered to subscriber C.

- There are several possibilities at this point:
  - At least one of Subscriber C's UEs is ringing. Subscriber B hears ring back tone. If subscriber C answers the call, the UE on which the call was answered becomes the attendant, and subscriber B and subscriber C are connected. Afterward, subscriber B flashes the hook to join subscriber A. After joining the call with subscriber A, it becomes a 3-way call. Subscriber B can then hang up to leave subscriber A and subscriber C on the call.
  - If subscriber C's phone is ringing and subscriber B flashes the hook flash. This joins the call with subscriber A that is on-hold. If C's phone is still ringing, subscriber A can hear the ringing after subscriber B flashes the hook for the second time.
  - If subscriber C answers the phone, subscriber B and subscriber C are able to talk to each other. If subscriber C does not want to accept the CT and hangs up the phone, Subscriber B can press the hook flash to return to subscriber A that is on hold.
  - If subscriber C's phone is ringing, subscriber B hears the ring back tone. If there is no answer, the result is the same as in the case of a Blind Call Transfer.
  - If subscriber C's phone is busy, subscriber B can flash the hook to get back to subscriber A. In that case, the connection to subscriber C is terminated.

Service providers can cause any charges for transferred calls to be assigned to Subscriber A, B, or C.

This feature is described in detail in [GR 579]. Please consult [GR 579] for details about call termination and call failure scenarios.

In the case where C has multiple UEs registered on its public identity, the UE that B is talking to is the same UE that A is transferred to, in the above example.

### **6.5.3.3 Call Transfer Restrictions**

There can be restrictions on an incoming call, restricting the addresses it can be transferred to or the addresses an outgoing call can not be transferred to. Possible restrictions on CT are as follows:

- The subscriber cannot transfer a call to a charged call, such as 9xx or a collect call.
- The subscriber cannot transfer a call to an international call or long distance call.
- The subscriber cannot transfer a call to a destination in the PSTN, or outside of the service provider's domain. This may impact billing and service infrastructure.
- The subscriber cannot transfer a call during certain dates and times. This may be due to billing issues or curfews.

### **6.5.4 Three-Way Calling (3WC)**

Three-Way Calling (3WC) allows a subscriber to add a third party to an existing two-party conversation without operator assistance.

A description of this feature is given below:

- To start a three-way call, a UE should be involved in a stable two-way call (it does not matter which party originated the call).
- The subscriber places the first call on hold by pressing a conference call button; if there is no conference call button, a hook flash works. The UE puts the first call on hold, and apply the stutter tone followed by the dial tone.
- After receiving the dial tone, the subscriber dials the phone number of the third party.

- To join the two call legs together thereby establishing the three-way call, the subscriber press the conference call button again. If there is no conference call button, the subscriber should flash the hook again. Upon the hook flash or pressing conference call button, the second party joins the call. This can occur while the subscriber is receiving audible ringing for the second call or after the third party answers the call. The subscriber may speak privately with the third party before sending the second flash.

The analog user interface for Three-Way Calling is described in [GR 577] and [GR 578]. Please consult these documents for details on call termination and failure scenarios.

The Three-Way Calling Feature is made available to a subscriber through UE provisioning by the service provider.

## 6.6 Call History Features

### 6.6.1 Auto Recall (Last Call Return)

Automatic Recall (AR) allows a UE registered with a Public Identity to automatically call back the last calling address (that target address is the P-Asserted-ID of the caller) that sent an INVITE to this UE, whether the INVITE was answered by this UE or not. The AR feature should work even when the last call received at the UE did not supply a caller ID.

A detailed description of the user interface specified by Telcordia may be found in [GR 227]. This is the desired user interface for UEs that provide an analog interface to an analog telephone-like device. The following is a summary of that, with some adjustments to terminology:

- A subscriber may activate AR at any time by going off-hook, receiving dial tone, and dialing the AR feature activation code. The suggested activation code is \*69. The AR user hears a call attempt to the target address, and if the target is busy, the AR user hears an announcement saying that AR has been activated and if the target becomes available, your phone shall ring.
- If the Target is busy, then the busy/idle status of the called and AR UE's status is monitored periodically. If both the AR UE and Target are found idle before a timeout occurs, the AR UE is given special ringing.
- If the AR requesting user does not answer the special ringing within a specified number of ringing cycles, monitoring of the Target user equipment resumes. If the AR requesting user answers the special ringing, call setup is attempted.
- If the Target user equipment becomes busy after the AR UE answers special ringing, then the user hears the resume scanning announcement and monitoring of the Target resumes. If the Target is idle after the AR UE answers special ringing, then the user hears audible ringing and the AR request is completed.
- The AR requesting user may deactivate AR at any time by going off-hook, receiving dial tone, and dialing the appropriate deactivation code. The suggested service deactivation code for AR is \*89. Once the deactivation code is dialed, the user hears an announcement stating that all outstanding AR requests have been deactivated.
- A user may reactivate an AR request by dialing the AR feature activation code if the last call received by the Public Identity is associated with an existing AR request. A reactivation increases the duration of time that the Target is to be monitored.

A client with a richer and/or different user interface may provide a different user experience, providing comparable functionality.

As described in [GR 227], the service provider has the option of one-stage or two-stage activation of AR. Two-stage activation announces the number to be recalled and asks for confirmation from the user before activation. Two-stage activation is not necessary in the PacketCable environment, meaning that the voice-back option is optional, and its use is provisionable by operators.

### 6.6.2 Auto Callback

Automatic Callback (AC) allows a UE registered with a Public Identity to request automatic initiation of call set up to the address last called by the requesting UE (whether the call was answered or not), without the need for a user to redial the address. This feature is directed at users who need to reach a party that is currently busy. This feature may also be used to call back the last party that was called, even if the called party's line is not busy. The AC feature should work even when the last call did not supply a caller ID. This happens when the last call from the AC requesting UE is an anonymous AR call.

The AC functionality is substantially similar to that of the AR feature, except that the two-stage dialing option is not available for the AC feature. A detailed description of the user interface specified by Telcordia may be found in [GR 215]. This is the desired user interface for UEs that provide an analog interface to an analog telephone-like device. The following is a summary of that, with some adjustments to terminology:

- A subscriber may activate the AC feature at any time by going off-hook, receiving dial tone, and dialing the AC feature activation code. The suggested activation code is \*66. Once the subscriber has dialed the AC feature access code, the subscriber hears one of the following:
  - An audible ring indicating that call setup is being attempted immediately (that is, Immediate Processing).
  - A "delayed processing" announcement indicating that the Target is busy and that the system attempts to complete the call once the Target is idle.
  - A short-term denial announcement.
  - A short-term denial tone.
  - A long-term denial announcement.
  - A denial announcement indicating that the called party has a call rejection feature active and is not accepting calls from the requesting AC UE. The denial announcement is provided by the called party's call rejection feature as a result of call setup after AC processing. The announcement is given, for example, if the called party has Selective Call Rejection active for calls from the calling party, the called party has Selective Call Acceptance with the denial announcement option active and is not accepting calls from the calling party, or the called party has Anonymous Call Rejection active and the calling party's address is marked anonymous. Another example of an AC user receiving a denial announcement is when the called party has DND active.
- If the Target is busy and the AC UE has received the delayed processing announcement, then the busy/idle status of the Target and AC UE's status is monitored periodically. If both the AC UE and Target are found idle before a timeout occurs, the AC UE is given special ringing.
- If the AC requesting user does not answer the special ringing within a specified number of ringing cycles, monitoring of the Target resumes. If the user answers the special ringing, call setup is attempted.
- If the Target becomes busy after the AC UE answers special ringing, then the user hears the resume scanning announcement and monitoring of the Target resumes. If the Target is idle after the AC UE answers special ringing, then the user hears audible ringing and the AC request is completed.
- The user can deactivate AC at any time by going off-hook, receiving dial tone, and dialing the appropriate deactivation code. The suggested service deactivation code for AC is \*86. Once the deactivation code is dialed, the user hears an announcement stating that all outstanding AC requests have been deactivated.

A client with a richer and/or different user interfaces may provide a different user experience, providing comparable functionality. For instance, the number to be recalled may be displayed rather than announced.

## 6.7 Miscellaneous Features

### 6.7.1 Do Not Disturb Client (DND Client)

**Note:** This feature is out of scope for the first phase of RST specification, where it is assumed that there is a single client behind a given public identity.

The Do-Not-Disturb client feature allows the user to put a given UE into a mode in which it does not alert on receipt of incoming calls. This feature is distinct from the Do-Not-Disturb public identity feature in that it affects only a single UE, not all UEs behind a single public identity.

The user enables the feature through some interaction with the client. From the time the feature is enabled until it is disabled (either by user interaction or by some error condition such as client power loss), the client does not alert. The activation state for DND Client persists as long as the client has power.

In cases where there are multiple UEs per public identity, and a call is offered to the public identity, several responses might be sent for each of the UEs registered to the public identity. The UE with DND Client active would send an indication of DND, however, another UE behind that same public identity may answer the call. The UE or proxy receiving these responses should pick the "best" response with a goal of connecting to something.

If there are two or more UEs behind a single public identity and only one UE has DND Client activated when a call arrives for the public identity, all UEs should alert except for the UE with DND.

Only the UE that activated DND Client can deactivate DND Client.

When the DND feature is activated, the default behavior (without considering feature interactions and forking of calls to multiple UEs) is for incoming calls to be declined by the UE and the network plays a special DND greeting notifying callers that the party they have called is not receiving calls at that time. Callers may have the opportunity to enter a PIN to override the Do-Not-Disturb-Client feature. If the correct PIN is entered the call is offered to the UE again with an indication that Do-Not-Disturb-Client is overridden. DND subscribers can use the SPP feature to program the DND override PIN.

Certain callers can override the DND Client feature, including emergency services operators and operator services operators.

In the case when an incoming call has been forked to more than one UE, only one of which has Do-Not-Disturb-Client activated, the UE that has Do-Not-Disturb-Client activated declines the call.

In the case where there is only one UE registered for a public identity, and DND-Client is activated, the feature behavior is equivalent to DND-Public-Identity.

The DND Client feature is activated and deactivated via a per UE-type interaction with the subscriber. UEs with analog user interfaces could use a "star-code" to activate and deactivate the feature. For those UEs with analog interfaces, when the feature is activated there is a feature activation confirmation, so that the user knows the feature was successfully activated.

When DND Client is activated there should be a status indicator at the client, either visual or audible when off-hook, that the DND feature is activated.

### 6.7.2 Do Not Disturb Public Identity (DND Public Identity)

**Note:** In the first phase of RST specification, it is assumed that there is a single UE behind a given public identity. In this case, the DND Public Identify feature is simply called the DND feature.

The Do-Not-Disturb public identity feature allows the user to put the public identity and all UEs behind it into a mode in which they do not alert on receipt of incoming calls. This feature is distinct from the Do-Not-Disturb client feature in that it affects all UEs behind the public identity, not just the UE that activated the feature.

The user enables the feature through some interaction with a client. From the time the feature is enabled until it is disabled, callers receive an indication that the called party is not available to receive calls and that none of the clients alert. The activation status of the DND Public Identity persists even if the client that registered the last update loses all power.

When the DND feature is activated, the default behavior is for incoming calls to be declined by the network, which plays a special DND greeting notifying callers that the party they have called is not receiving calls at that time. Callers may have the opportunity to enter a PIN to override the Do-Not-Disturb-Client feature. If the correct PIN is entered the call is offered to the UE again with an indication that Do-Not-Disturb-Client is overridden. DND subscribers can use the SPP feature to program the DND override PIN.

Certain callers can override the DND Public Identity feature, including emergency services operators and operator services operators.

There should be an optional indicator (based on service provider policy) that DND Public Identity is active, provisioned through the configuration interface to all the UEs registered to the public identity.

The client or clients that are registered to a public identity that has DND Public Identity activated are not offered calls, so there can be no client-based call history logging of calls while DND Public Identity is activated.

The DND Public Identity feature is activated and deactivated via a per UE-type interaction with the subscriber. Clients with analog user interfaces could use a "star-code" to activate and deactivate the feature.

### **6.7.3 Screening List Editing (SLE)**

The Screening List Editing feature allows users to create and update lists of public identities (in the form of the Directory Number, Tel or SIP URI) associated with features that require these lists, such as Selective Call Forwarding, Distinctive Alerting, Do Not Disturb and Solicitor Blocking. This feature also allows the user to activate/deactivate features that use lists and obtain feature status reports.

For an UE with an analog telephony interface, the general behavior of the SLE feature (especially its user interaction) is described in [GR 220].

For an UE that does not have an analog telephony interface, the user interaction procedures for the SLE feature can be designed to take advantage of the specific user-interface capability (e.g., a web client) provided by the UE.

### **6.7.4 Subscriber Programmable PIN (SPP)**

The subscriber programmable PIN feature allows a subscriber to manage PIN codes for features that require personal identification numbers. When a subscriber is given access to a feature that requires a PIN, the service provider provides an initial PIN. This PIN must be changed the first time that the subscriber attempts to use the PIN. The SPP feature provides an interface for the subscriber to change PIN codes for different features on an as needed basis (i.e., because they are being forced to change the initial PIN, or because they want to change their PIN for personal reasons).

The SPP feature is accessed from the public identity that is subscribed to the SPP feature. This helps to improve the security of the SPP feature by preventing someone other than the subscriber from changing the subscriber's PIN. SPP can be used to program PIN codes for the following features: RACF, DND override, SCF, SCA, and SCR. Subscribers can have different PIN values for different features.

If the subscriber attempts to use a service-provider-supplied PIN for the first time and the subscriber is calling from his public identity, the subscriber is prompted to enter a new "N" digit PIN number. If the subscriber enters a valid length PIN that is different from the PIN supplied by the service provider, the subscriber is then prompted to enter the same PIN again for verification. If the PIN numbers provided in the initial and confirmation attempts are the same, the new PIN is stored and the subscriber is notified of the success of the PIN change. If the two PIN attempts do not match, the subscriber is prompted to try again. The number of PIN programming attempts is limited to some service provider-provisioned value.

If the subscriber cannot successfully complete PIN programming within the allowed number of attempts, the subscriber is notified of the PIN programming failure, asked to try again later and the call is terminated.

If a subscriber desires to change a PIN that was not supplied by the service provider, they interact with the service provider's SPP interface. This interface can be a web programming interface or an interactive voice response system. Only the IVR interface is described in this document.

To change a PIN code, the subscriber calls the SPP feature from his own public identity. The SPP feature then prompts the caller to enter the feature code for the feature whose PIN is to be added or changed. For example, if the DND PIN is to be changed, the subscriber would enter the DND access code. If the subscriber enters an invalid feature access code or a valid feature access code, but for a feature that he is not subscribed to, the feature prompts the user to re-enter the feature access code. If the subscriber fails to enter a valid feature access code in the allowed number of attempts, the subscriber is notified of the SPP programming failure and the call is ended.

Once a valid feature access code has been entered, the SPP feature announces the name of the feature associated with the access code and the subscriber is prompted to enter the existing PIN for that feature. If the subscriber enters a PIN that doesn't match the stored PIN, the subscriber is given the chance to retry with another PIN. If the subscriber fails to enter a valid existing PIN in the allowed number of tries, the subscriber is notified of the SPP programming failure and the call is terminated. If a valid PIN is entered for the existing PIN, the subscriber is prompted to enter a new PIN. If a valid length PIN is entered that is different from the existing PIN, the subscriber is then prompted to enter the same PIN again for verification. If the PIN numbers provided in the initial and confirmation attempts are the same, the new PIN is stored and the subscriber is notified of the success of the PIN change. If the two PIN attempts do not match, the subscriber is prompted to try again. The number of PIN programming attempts is limited to a service provider provisioned value.

The SPP feature should generate a log each time the number of existing PIN entry retry attempts is exceeded.

If any of the following conditions occur during an SPP programming attempt, the caller gets a failure indication and the call is terminated:

- Subscriber fails to enter a valid existing PIN in the allowed number of attempts.
- Subscriber fails to enter a valid feature code in the allowed number of attempts. A valid feature code is defined as a feature code that identifies a feature that both uses PIN codes and is subscribed to by the caller.
- Subscriber fails to enter a valid new PIN code. A valid new PIN code is one in which the PIN consists of the provisioned number of digits and one that does not match the existing PIN.
- Caller is not subscribed to SPP.

**Note:** This mechanism for entering PIN is one of several options available for PIN programming, and may not be required for a vendor implementation, depending on operator requirements.

#### 6.7.5 Distinctive Alerting

UEs are user-configurable or provisionable to provide customized alerting behaviors (audible, visual, vibration, etc). The behavior used in any given condition:

- May be selectable through feature definition on the part of an operator.

- May be selectable by end-user configuration.
- May be selectable based on received call signaling, at a minimum based upon the public identity that is called and/or the public identity that is calling, by the UE and/or by the network.

Media content used in alerting behaviors may be stored locally on the UE or a reference (such as a URL) to a remote source may be used.

Media content may be created by the operator, by third parties (if permitted by the operator), or by the user (if permitted by the operator).

When Caller ID is blocked for the calling party and the distinctive alerting feature is triggered based upon the identity of the calling party, distinctive alerting is performed if and only if this feature has been configured to take precedence over Caller ID Blocking feature.

### 6.7.6 Message Waiting Indicators

Message Waiting Indicators are presented when a message has been received by the network for the public identity associated with the UE. Traditionally a message is a voice mail, but may be extended to additional message types based on additional to-be-defined feature definitions.

In the case of traditional voice mail, the Message Waiting Indicator capability is invoked in response to a notification that a voice mail message has been received. The UE must be provisioned for a voice mail service in order to receive these notifications.

Visual Message Waiting indicator presentation options range from a simple light that is illuminated whenever any message has arrived in the network for the public identity, to an alphanumeric display that provides information on the number of messages that are waiting in various categories.

For user interfaces that provide analog audio indications that a message is waiting, the usual indication is a stutter dial tone that is supplied when the phone is off-hook.

When messages have been received by the subscriber, the message waiting indicator is removed by the UE.

### 6.7.7 Speed Dialing

Speed Dialing allows one to dial frequently-called numbers by assigning them either a one-digit or two-digit abbreviation code. One-digit speed dialing usually accommodates eight numbers (2 - 9) and two-digit speed dialing can accommodate up to 99 numbers, but is usually limited to a more reasonable set of frequently-dialed numbers (e.g., 30 numbers). If both one- and two-digit dialing are allowed simultaneously, then different feature access codes are needed for each (e.g., \*74 for one-digit speed dialing and \*75 for two-digit speed dialing).

After the feature access code is dialed, the one- or two-digit number representing a previously stored directory number (DN) is entered and the call is placed as if the entire DN was dialed.

Various schemes would allow the programming of the frequently-dialed director numbers. The numbers could be stored via a sequence dialed via the UE or entered via a web-based subscriber management function that allows for the creation of the list of numbers.

Two options would be available to the service provider concerning the network-maintained list of numbers: 1) one per public identity (shared list), 2) one per UE associated with the public identity (private list). With this functionality, the importance of web-based management of speed dialing and other subscriber-based features become critical.

**Note:** In the first phase of RST specification, it is assumed that there is a single UE behind a given public identity; in this case, these two options can be considered as the same.

It is assumed that this feature availability would be provisioned at subscription and stay in effect until the feature was removed from the subscription.

Refer to [GR 570] for more detail on the PSTN equivalent feature.

### 6.7.8 Customer-Originated Call Trace

The Customer-Originated Call trace (COT) feature allows the receiver of an obscene, harassing, or threatening call to initiate a call trace. This terminating call feature is often called Malicious Call Tracing, but the COT adds the capability of the offended party to invoke the trace. After the offended customer invokes the call trace procedure, the appropriate service authorized agency handles the disposition of the malicious call.

After receiving and ending the malicious call, the COT customer activates the feature by going off hook, listening for a dial tone, and then dialing the call trace activation code (usually \*57). The COT customer needs to activate the feature immediately after receiving the offending call, and prior to accepting a new call.

The COT feature simulates, where possible, the behavior defined in [GR 216].

For the case in which multiple UEs have registered using the same public identity, and the COT feature is invoked from one of these UE, the call that is traced is the last call answered and ended from any UE registered to the public identity from which the COT feature was invoked.

## 6.8 Fax, Voice-Band Data, and Other Media Types

### 6.8.1 Fax, Modem, and TDD Calls

UEs with analog interfaces should support origination and termination of fax, modem, and telecommunications device for the deaf (TDD) calls. Fax, modem and TDD calls may be transported in pass-through mode over G.711. Fax calls can also use T.38 fax relay ([ITU-T T.38]) to cope with networks that have packet loss. Modem calls can also use [ITU-T V.152] to cope with networks that have packet loss.

Pass-through mode is simply providing a G.711 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).

T.38 fax signaling is a mechanism to communicate fax data over TCP, UDPTL, or RTP with better reliability than that provided by pass-through mode. This is done by converting the outgoing signaling into T.38 data packets and encapsulating both fax signaling and fax data packets in RTP, UDPTL or TCP wrappers. The far-end can then interpret incoming T.38 IP packets and reproduce the analog signaling locally rather than relying on the G.711 pass-through of signaling tones, etc. For additional information on T.38 fax relay, please refer to [ITU-T T.38].

UEs may offer additional capabilities beyond PSTN analog line emulation. Opportunities to communicate with IP aware fax, modem, and TDD devices without the need to implement the final voice band analog translations exist. Definition of non-analog features and requirements to support such communications is not in scope for this document version.

If a public identity's primary use is for data calls, it should be possible to configure the public identity such that operator interrupt is disallowed. This prevents an operator interrupt from disrupting data transmission.

### 6.8.2 Home Alarm System Support

Home alarm systems may use 2-wire loop start telephone interfaces as an integral part of an alarm service. Example home alarm system use of the telephone interface includes:

- Reporting an intruder to an alarm center via dial-pulse or DTMF signaling.

- Detecting analog loop current loss.
- Transmitting a heartbeat signature to an alarm center.
- Supporting remote software upgrades.

Legacy home alarms systems assume voice band communications and DC battery feed performance are compliant with traditional PSTN networks. This has resulted in non-standardized voice band communications and signaling techniques that require coherent transmission of the entire voice band. It is recommended that UEs with analog interfaces that connect to alarm systems be configured to negotiate a SIP session supporting G711 A/MU law codec that coherently transmits the 300 – 3400 hertz voice band. To avoid false alarms at the home alarm system, it is also recommended that UEs embedded in cable modems with analog interfaces that connect to alarm systems avoid removing DC Bias from the analog interface unless they lose DOCSIS connectivity.

### 6.8.3 DTMF Relay

UEs with analog interfaces should support dual-tone multi-frequency (DTMF) detection, demodulation, network transport, and regeneration. UEs may negotiate to use [RFC 2833] as the means to transport DTMF signals.

[RFC 2833] specifies in-band RTP payload formats and usage to carry DTMF, modem and fax tones, line states, and call progress tones across an IP network either as recognized "telephone events" or as a set of parameters defining a tone by its volume, frequency, modulation and duration of its components.

Besides the transport of tones across an IP network, [RFC 2833] also allows for the remote collection of DTMF digits by a media gateway to relieve an Internet end system (e.g., media server) of having to do this. Other advantages of [RFC 2833] include inherent redundancy to cope with packet loss and the means to allow IP phones to generate DTMF digits when signaling to the PSTN without requiring DTMF senders.

### 6.8.4 DTMF in Audio

UEs with analog interfaces should be able to accurately transmit or receive DTMF digits during an audio session. If the audio session uses a compression codec that is not capable of accurate DTMF pass-through, [RFC 2833] should be used for DTMF relay.

## 6.9 Early Media

Early media refers to the exchange of media (such as audio) in the context of session even before the called party answers the call. Since "answer" is a billing concept that often depends on call/service state, this section attempts to define early media as distinct from the concept of answer. For this initial issue of the PacketCable Residential SIP Features technical report, the term media is restricted to mean audio.

Early media flows are intended to receive the similar QoS treatment to normal media packet flows.

### 6.9.1 One-Way Early Media

One-way early media is audio delivered to a caller after the sending of an INVITE and before the caller receives a final response to the INVITE.

Some examples of one-way early media are:

- Delivery of an announcement stating that the call has failed, and explaining why.
- Delivery of an in-band ring back tone.
- Media sent by the callee after answering the call, which happens to reach the caller before the answer signaling.
- Interactions with PSTN calls that expect to receive in band audible ringing tone.

## 6.9.2 Two-Way Early Media

Two-way early media is audio exchanged in both directions between caller and callee after the sending of an INVITE and before the caller receives a final response to the INVITE.

Some typical examples of two-way early media are:

- A subscriber calls someone who has enabled the Do Not Disturb feature. The caller is prompted to enter a PIN code before the call is allowed to proceed.
- The caller gets ring-no-answer or busy and the network announces that for a nominal fee it will complete the call for the caller and the caller needs to press a key to activate the feature.
- A prepaid service, where the caller must interact with an IVR to provide a PIN and destination before the call can be completed.
- Voice activated calling – the subscriber picks up the phone, receives a prompt to speak the name of a destination and it is dialed for the subscriber.
- PSTN interactions for features such as credit card calling.

## 6.10 Operator Services

PacketCable Residential SIP Telephony services require an operator services platform to be able to verify that a line is busy and interrupt a call in progress. The operator services platform may or may not use UEs for its operator stations, and therefore may be using traditional circuit switched components, so the specification for operator services functions shall provide options for interoperating with a circuit switched operator services platform.

In the Busy Line Verification Feature (BLV), a subscriber calls an operator services position to ask the operator to determine if a line is in use. In Operator Interrupt Service, the operator provides a Busy Line Verify Service and then interrupts a call in progress and relays a message from the requester of the interrupt to the interrupted party. If the interrupted party is willing to hang up, the call can then be placed by the caller to the called party. If the caller requests the feature from the operator, the connection between the caller and the interrupted party can be reinitiated by the operator using the Call Completion Feature after the called party hangs up.

A conventional North American telephone environment requires a dedicated BLV trunk between the operator services system and the core telephone system. These trunks typically use Multi-Frequency tone-based signaling.

Busy Line Verification, Operator Interrupt, and Call Completion Feature shall follow the requirements specified in [GR 1176] FSD 80-01-0300 and FSD 80-01-0500. Verification connections are described in [GR 531].

### 6.10.1 Busy Line Verification

Requesters access the Busy Line Verification feature by dialing 0+[NPA] NXX-NXXX. [NPA] NXX-NXXX is the line to be verified. This connects the requester to the operator services position. The requester asks the operator to perform the Busy Line Verification. The line to be verified (the target line) may be directly accessible by this operator services position. In that case, the operator services position first calls the target line to verify that it is busy. If the line is not busy (it rings) or if the call cannot be completed due to some non-busy type of failure returned by call processing signaling, the operator informs the subscriber of the state of the called line. If the line is busy, the operator position bridges into the target's call in progress (in listen-only mode) and listens to the called line with automatic scrambling of the called line speech (in both the upstream and downstream direction) to ensure privacy while the operator listens for conversation. This scrambling is provided as a local function to the operator services position.

Lines may be provisioned to deny busy line verification in order to protect digital data transmission. In that case, the operator notifies the requester that busy line verification is not possible.

If the target of a BLV request has more than one UE behind a public identity (as may be possible in future iterations of PacketCable Residential SIP Telephony features), and at least one UE is not in use, then the public identity is not considered busy, so the operator cannot verify the public identity is busy and therefore cannot interrupt the caller either.

### 6.10.2 Operator Interrupt

Operator Interrupt is performed as a follow-on step after the Busy Line Verification feature. The requester asks the operator to interrupt the call. Operator Interrupt is allowable during the addressing state (dialing), the call-processing state (waiting for a call to complete), and the communications state (talking in a call) as defined in [GR 506]. Lines may be provisioned to deny Operator Interrupt. In that case, the operator notifies the requester that Operator Interrupt is not possible. If the feature is allowed, the operator services position removes the scrambler from the line and injects an intrusion tone such as a 440 Hz Call Interrupt tone into the call and interrupts the called party's conversation. The connection from the operator is re-configured from listen-mode to 2-way talk mode, and the other connection(s) is transitioned to inactive so that only the target party should be able to hear the operator. If the called party is willing to establish conversation with the caller, the called party can hang up to allow the caller to re-originate the call.

If the caller desires Operator Interrupt followed by Call Completion, the operator informs the caller of the verified line's status and whether the interrupted party is willing to establish conversation. If the interrupted party is willing to establish conversation, the operator services system then connects the calling and called parties. If the interrupted party refuses to establish conversation, the call is not completed. With Call Completion, the operator connects the calling party to the called party directly without routing the call through the operator services platform.

## 6.11 Emergency Services

When initiating a session to an emergency services address, a UE supplies its current location information within the signaling for the session, in order to allow the call routing network elements to direct the call to the correct PSAP and aid emergency services personnel in providing emergency services to the initiator.

### 6.11.1 Assumptions

In the current version of this document, the following assumptions have been made:

- The location of UEs is subject to change across registrations.
- Support for emergency services requires that a UE has been registered and authenticated before an emergency call is placed.
- UEs store their location information along with the time that location was determined.
- Optional prioritization of emergency services sessions is based on service provider policies.

### 6.11.2 Emergency Calling

A user initiates an emergency session by using a UE to initiate a session to an emergency address. For analog line clients, this is performed using traditional dialing of emergency services telephone numbers, such as "911" in North America.

Detection and routing of emergency services function independently of the underlying IP connectivity network.

When an emergency call is placed, the UE tags the call signaling to indicate that the call is an emergency services call. The UE recognizes any of the emergency services phone numbers or addresses currently in use in the jurisdiction in which the user subscribed for service. The PacketCable network then has the option of prioritizing emergency sessions over "ordinary" sessions, according to the prevailing policy of the service provider.

In order to make every effort to deliver an emergency call to an emergency operator, if the PacketCable network receives a call targeting an emergency services telephone number or address, and the call is not tagged as an emergency services call, the network still attempts to complete the call to the emergency service operator.

A UE that is provided dial tone can place an emergency call.

The emergency center or PSAP targeted by an emergency services call may be connected to any kind of voice network. When the PacketCable network routes the emergency services call to the emergency center or PSAP, the network supplies the location of the caller or at least, it should supply enough information for the PSAP to look up the caller's location. The Emergency center or PSAP targeted by an emergency services call is able to call back the UE for an emergency session from a UE that has registered that contains valid credentials.

The PacketCable network routes emergency calls to the correct PSAP based on originator's location.

Features such as hold, conference, transfer, and call-waiting are disabled during emergency calls.

Once an emergency session is established between a caller and an emergency operator, the operator can prevent the release of the call, even if the caller hangs up. The operator can also attempt to ring-back a caller if the caller is unresponsive, whether the caller's phone is on-hook or off-hook.

When an emergency operator ring-back is requested, the operator is allowed to cause the caller's phone to performed timed ringing (if it is on-hook) or to play Receiver-Off-Hook tone toward the user (if the phone is off-hook).

### **6.11.3 Location Identification**

UEs store their location based upon at least one of the following methods:

- UEs not behind NAT are informed of their location through DHCP.
- UEs may automatically determine their location (for example, through GPS).
- The network may automatically determine UEs location (for example, through triangulation).
- Subscribers may self-report their location to the operator, and the operator may provision the location in their network.

### **6.11.4 SIP Location Conveyance**

UEs convey the device location information, if known, in the SIP signaling with emergency calls as defined by standard SIP location conveyance mechanisms.

## **6.12 Monitoring and Management**

### **6.12.1 In-Service and Out-of-Service States**

In the PSTN, dial tone indicates to the caller that he or she is connected to the network (in reality, connected to the line card of the Class 5 switch). The many years of PSTN experience have led us to believe that no dial-tone always means that the telephone is out-of-service. UEs, however, have the intelligence and capability to generate the majority of call progress tones locally, including dial tone. So dial tone could be provided regardless of the network connectivity status.

Normal UE operation would call for the registered UE to locally generate dial-tone, then collect dialed digits and use a previously provisioned digit map that directs the UE when to send and how to form an INVITE to the network to begin establishing the desired session. Any success or failure indication would not occur until after the initial INVITE was sent. If the UE was truly "unconnected" to the network, the UA could only interpret this when no SIP

response message was received from the INVITE. This would occur much later in the dialing sequence and provides a significant departure from PSTN behavior.

The in-service and out-of-service feature described here provides UEs with the ability to reproduce a similar service representation to subscribers as the one used in the PSTN: the dial-tone indication is provided to the user only if the UE is in-service. That is, a dial-tone indicates that the UE is successfully registered with the PacketCable network, furthermore it is able to initiate sessions to the network (at least to the access proxy). Thus, there is no behavioral shift required for users moving from traditional PSTN to PacketCable services.

### 6.12.2 VoIP Metrics

UEs can monitor, process, and report key voice quality metrics that indicate the voice quality of the sessions. PacketCable has identified five areas of interest:

- Configuration – Configuration data that affects voice quality (e.g., jitter buffer size) are reported.
- Packet Loss – Packet loss percentage is reported which may indicate link failure, congestion, or miss-routed packets.
- Jitter – Jitter, the packet-to-packet delay variation, may cause packet loss when the jitter buffer is overrun.
- Delay – High levels of delay causes problems with normal conversation.
- Echo – Hearing your own voice after a slight delay also causes problems with normal conversation.

Statistics reporting includes a unique correlation identifier to allow an operator to correlate statistics with information available from other network elements that were involved with the call.

## 6.13 Accounting

This section lists the various Residential SIP Telephony call features and determines if call accounting is necessary. Since this section is discussing network usage and accounting within the scope of Residential SIP Telephony, the focus is on events that occur during traditional telephony that emulate many of the PSTN features.

The nature of UEs introduces several potentially complicating factors for tracking network usage and accounting:

- UEs that are not embedded with the cable modem are not considered a trusted source for accounting data. UEs are expected to handle more of the call processing than PacketCable 1.5-based MTAs, so care needs to be taken around procedures that rely on the UE as a source for accounting data.
- SIP Servers (i.e., CSCF or AS) are expected to be the source for most of the accounting data, such as timestamps, resource usage, originator and terminator.
- In a distributed voice network architecture, accounting events may need to be captured from several different network elements.
- The UE may be one of several registered to the same public identity, which can result in additional information that should be captured for accounting purposes. For this initial issue of the Residential SIP Telephony Feature Technical Report, the case of multiple UEs behind a single public identity is not considered.

Table 1 below lists presently defined call features within Residential SIP Telephony and whether call accounting records may be required.

**Table 1 – Accounting for Call Features within Residential SIP Telephony**

<b>Residential SIP Telephony Feature</b>	<b>Accounting Data Source</b>	<b>Notes</b>
Basic Call	SIP	Feature usage based accounting required.
Caller ID	None	Provisioning based accounting only. No usage based accounting required.
Call Forwarding	SIP	Feature usage based accounting required.
Call Blocking	SIP	Feature usage based accounting required.
Call Waiting	None	Provisioning based accounting only. No special Call Waiting usage based accounting required. Treated like a Basic Call for accounting.
Hold	None	
Call Transfer	SIP	Feature usage based accounting required.
Three-Way Calling	SIP	No special accounting required. Treated like 2 separate Basic Calls for accounting.
Auto Recall (Last Call Return)	SIP	Feature usage based accounting required.
Auto Callback	SIP	Feature usage based accounting required.
Do Not Disturb Client	None	Provisioning based accounting only. No usage based accounting required.
Do not Disturb Public Identity	None	Provisioning based accounting only. No usage based accounting required.
Screening List Editing	None	
Subscriber Programmable PIN	None	
Distinctive Alerting	None	
Message Waiting Indicators	None	
Speed Dialing	SIP	Feature usage based accounting required.
Customer-Originated Call Trace	SIP	Feature usage based accounting required.
Fax, Voice-Band Data, and Other Media Types	SIP	No special accounting required. Treated like a Basic Call for accounting. SDP details are provided in accounting records.
Early Media	SIP	Handled as a normal part of the feature that invokes Early Media.
Operator Services	SIP	Feature usage based accounting required.
Emergency Services	SIP	No special accounting required. Treated like a Basic Call for accounting.

## 6.14 Lifeline Capabilities

Lifeline capabilities of a RST CPE device enable the user to obtain sufficiently reliable and guaranteed service, even during utility power failure at the user's premises. These capabilities allow the user to have a similar experience with the device as with a PSTN-connected phone in terms of its availability particularly during utility power outage.

The RST specifications focus on the requirements on the CPE device's capabilities of sustaining the Residential SIP Telephony service when an outage occurs to the utility AC power supply. Capabilities of other network components are not addressed.

The lifeline CPE device contains a battery that provides auxiliary power during interruption of AC Power supply. It generates alarms whenever AC Failure is detected, when the battery needs replacement, when the battery is missing, and when the battery is low (for example, 75% discharged). To ensure that the service remains available during utility power outages, the lifeline CPE device is equipped with an integrated uninterruptible power supply (UPS). Such a device also has LEDs that show the battery status on the device itself. In general, a lifeline CPE device has the same power and battery capabilities as were defined for E-MTAs defined in [AIP1.5].

During AC power outage, all lines, telephone extensions, and UEs associated with a lifeline device with battery power should remain operational, so long as the AC Power outage is not affecting the cable plant. In particular, subscribed Residential SIP Telephony features should remain available to users of the lifeline device according to the service provider's provisioning.

## Appendix I User Interface Tones and Signals

PacketCable Residential SIP Telephony 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
	DI	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	Ringling	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:

1. 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.
2. 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.
3. 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.

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