PacketCable™

Residential SIP Telephony Feature Specification

PKT-SP-RSTF-I01-060927

ISSUED

Document Status Sheet

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Closed	A static document, reviewed, tested, validated, and closed to further engineering change requests to the specification through CableLabs.

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1 SCOPE

1.1 Introduction and Purpose

This document specifies an implementation of common residential telephony features in a PacketCable network with SIP based User Equipment (UEs). These features include the most commonly used residential features and capabilities, including Caller ID features, call forwarding features, hold, transfer, three-way calling, emergency calling, and operator services.

The PacketCable network architecture uses the Session Initiation Protocol (SIP) as the basis for call setup and teardown. SIP was designed to permit distribution of feature logic to the network endpoints, requiring less processing in network elements, and theoretically permitting a reduction in cost of network infrastructure. Also, SIP was designed to support a wide range of interactive applications, including telephony, video conferencing, interactive messaging, presence, wireless services, and many others. Lastly, SIP is the foundation of the IP Multimedia Subsystem (IMS) architecture, upon which the PacketCable architecture is based. Many telecom equipment and software suppliers are focusing their investments in converged network solutions on the IMS architecture, and the PacketCable architecture leverages the IMS architecture, which should yield more rapid implementation timeframes.

This initial Residential SIP Telephony Feature Specification is intended to apply to User Equipment (UE) that interfaces to an analog phone.

1.2 Requirements

Throughout this document, the words that are used to define the significance of particular requirements are capitalized. These words are:

"MUST"	This word means that the item is an absolute requirement of this specification.
"MUST NOT"	This phrase means that the item is an absolute prohibition of this specification.
"SHOULD"	This word means that there may exist valid reasons in particular circumstances to ignore this item, but the full implications should be understood and the case carefully weighed before choosing a different course.
"SHOULD NOT"	This phrase means that there may exist valid reasons in particular circumstances when the listed behavior is acceptable or even useful, but the full implications should be understood and the case carefully weighed before implementing any behavior described with this label.
"MAY"	This word means that this item is truly optional. One vendor may choose to include the item because a particular marketplace requires it or because it enhances the product, for example; another vendor may omit the same item.

2 REFERENCES

2.1 Normative References

In order to claim compliance with this specification, it is necessary to conform to the normative requirements within the following standards and other works as indicated. This is in addition to the other requirements of this specification. Notwithstanding, intellectual property rights may be required to use or implement such normative references.

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2.3 Reference Acquisition

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- Cable Television Laboratories, Inc., 858 Coal Creek Circle, Louisville, CO 80027; Phone 303-661-9100; Fax 303-661-9199; Internet: http://www.cablelabs.com /
- European Telecommunications Standards Institute, http://www.etsi.org/home.htm
- Internet Engineering Task Force (IETF), Internet: http://www.ietf.org/
  Note: Internet-Drafts are draft documents valid for a maximum of six months and may be updated, replaced, or obsoleted by other documents at any time.
  The list of current Internet-Drafts can be accessed at http://www.ietf.org/ietf/1id-abstracts.txt.
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- Telcordia Technologies, http://www.telcordia.com/

3 TERMS AND DEFINITIONS

This specification uses the following terms:

Active call / Active session	A call state where the called party has answered the call and two-way media is being exchanged.
Automatic Location Identification	The database that maps telephone number to location in the current 9-1-1 system.
Automatic Number Identification	The mechanism used to determine the telephone number of the caller.
Callback	When a PSAP initiates a new call back to a caller. This is not the same as ringback. See below.
CNAM	Calling Name.
Dynamic Host Configuration Protocol	The protocol used to configure an endpoint on an IP network commonly used to assign it an IP address, and recently extended to configure the location of the endpoint.
Emergency Services Gateway	A device which bridges the VoIP network to the Selective Router. In PacketCable this is a function of the MGC, SG and MG.
Emergency Services Query Key	A code that looks like a telephone number that is temporarily assigned to a VoIP 9-1-1 call by the VPC and is used as the key to the ALI database to retrieve location information for that call.
Emergency Services Routing Number	A code that is used to route a VoIP 9-1-1 call to the correct ESGW and also to choose an appropriate trunk group on that ESGW that connects to a specific Selective Router.
i2	The second 911 VoIP migration phase, called "i2" as defined by NENA providing a viable solution for VoIP carriers. There are two specialized "service operators" introduced in this migration phase: the VPC Service Operator and the Emergency Services Gateway Operator.
i3	The third and final migration phase planned by NENA, a more long term approach to address the needs of having IP-enabled emergency centers.
Integrated Services Digital Network	User Part (part of the SS7 signaling stack).
Internet Engineering Task Force	The standards body for the Internet.
Media Gateway	Devices bridging between the PacketCable IP Voice Communication network and the PSTN. A Media Gateway provides the bearer circuit interfaces to the PSTN and transcodes the media stream.
National Emergency Number Association	The primary standards body for the 9-1-1 community.
Off Hook	The active state of a traditional telephone, while a call is in progress or being attempted, and the telephone handset is out of its cradle.
On Hook	The idle state of a traditional telephone, while no call is in progress and the telephone handset is sitting in its cradle.
Proxy Server	An intermediary entity that acts as both a server and a client for the purpose of making requests on behalf of other clients.
Ringback	A function applied by a PSAP that causes an offhook phone to get ROH tone and an on hook phone to ring. This is not a new call. Ringback is not the same as Callback See above.

SIP Client	The functional element subscribers use to attach to the PacketCable network.
(ENUM) Telephone Number Mapping	In [RFC 3761], the E.164 to Uniform Resource Identifiers (URI) Dynamic Delegation Discover System (DDDS) Application (ENUM).
Trunk	An analog or digital connection from a circuit switch that carries user media content and may carry voice signaling (MF, R2, etc.).
VoIP Positioning Center	A specialized service operator who determines which PSAP should get a VoIP 9-1-1 call given the reported location of the caller and supplies that location to the PSAP when the PSAP consults the ALI.

4 ABBREVIATIONS AND ACRONYMS

This specification uses the following abbreviations:

3WC	Three Way Calling
AC	Auto Callback or Alternating Current
ACR	Anonymous Call Rejection
ALI	Automatic Location Identification
ANI	Automatic Number Identification
AR	Auto Recall
AS	Application Server
BLV	Busy Line Verification
CF	Call Forwarding
CFBL	Call Forwarding Busy Line
CFDA	Call Forwarding Don't Answer
CFV	Call Forwarding Variable
CID	Caller Identity Delivery
CIDB	Caller Identity Delivery Blocking
CIDS	Caller Identity Delivery Suppression
CNAB	Calling Name Delivery Blocking
CND	Calling Number Delivery
CNDB	Calling Number Delivery Blocking
COT	Customer Originated Trace
CPE	Customer Premise Equipment
CT	Call Transfer
CWT	Call Waiting
DHCP	Dynamic Host Configuration Protocol
DND	Do Not Disturb
DTMF	Dual Tone Multi Frequency
DVA	Digital Voice Adapter
E-CSCF	Emergency-Call Session Control Function
ENUM	Telephone Number Mapping
ESGW	Emergency Services Gateway
ESQK	Emergency Services Query Key
ESRN	Emergency Services Routing Number
GRUU	Globally Routable User-Agent URI
HSS	Home Subscriber Server
ICE	Interactive Connectivity Establishment
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
IP	Internet Protocol

ISDN	Integrated Services Digital Network
ISUP	Integrated Services Digital Network (ISDN) User Part
ITU	International Telecommunication Union
IVR	Interactive Voice Responder
MG	Media Gateway
MGC	Media Gateway Controller
NAT	Network Address Translation.
NCS	Network Call Signaling
NENA	National Emergency Number Association
OCB	Outbound Call Blocking
PACM	Provisioning, Activation, Configuration, and Management
P-CSCF	Proxy - Call Session Control Function
PIDF-LO	A Presence Information Data Format Location Object
PRACK	Provisional Acknowledgement
PSAP	Public Safety Answering Point – the entity that answers 9-1-1 calls
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RACF	Remote Activation of Call Forwarding
ROH	Receiver-Off-Hook, also known as "howler tone"
RST	Residential SIP Telephony
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol
SCF	Selective Call Forwarding
S-CSCF	Serving – Call Session Control Function
SDP	Session Description Protocol, defined by [RFC 4566].
SG	Signaling Gateway
SIP	Session Initiation Protocol, VoIP signaling protocol, defined by [RFC 3261]
SLE	Screening List Editing
SPP	Subscriber Programmable PIN
SS7	Signaling System #7
STUN	Simple Traversal of UDP through NATs
TCAP	Transaction Capabilities Application Part (part of the SS7 signaling stack)
TURN	Traversal Using Relay NAT
UE	User Equipment
URI	Uniform Resource Identifier
URL	Uniform Resource Locator
URN	Uniform Resource Name
UTC	Coordinated Universal Time
VM	Voice Mail
VoIP	Voice over Internet Protocol
VPC	VoIP Positioning Center

VSC	Vertical Service Code
XCAP	XML Configuration Access Protocol
XDS	XCAP Data Server

5 OVERVIEW

This Residential SIP Telephony Feature Specification is an application specification that relies upon the PacketCable architecture as described in [ARCH-FRM TR] and related PacketCable specifications. The Residential SIP Telephony specifications are embodied in a separate standalone release. This document specifies how residential telephony features must be implemented in PacketCable networks.

5.1 PacketCable Design Considerations for Residential SIP Telephony

The following design principles have been used in the creation of this specification.

- Minimal state awareness in network elements:

  This specification avoids holding state in network elements when it is reasonable to do so. The knowledge of any state related information is minimized in all network elements in order to favor scale and low cost implementations. Some examples of state information are: status related to the SIP and media sessions, device provisioning and QoS establishment.

- Distributed architecture for feature functionality:

  This specification distributes more functionality to end-user equipment when functional, security, and other requirements have allowed it. This creates more resilient and lower cost network solutions.

- Minimal changes to IMS specifications:

  IMS was designed to provide a network capable of supporting multiple applications; hence this specification avoids mandating changes to the IMS core network when it is reasonable to do so. This specification primarily defines additional requirements for User Equipment, Application Servers, and PSTN Interconnect elements such as Media Gateway Controllers.

- Transparency of end-user behavior for telephony service without inheriting the existing PSTN network architecture:

  As described above, the core architecture differs substantially from the existing PSTN network. Hence the behavior and feature interaction of some telephony features may differ from that described within Telcordia LSSGR specifications. In most cases the user behavior described within the referenced LSSGR specifications has been reproduced.

5.2 Residential SIP Telephony Specific Applications

This specification describes the set of requirements for the implementation of residential telephony features using a standard RST Client (called RST User Equipment (RST UE) or simply UE in this specification), which is in turn connected to a PacketCable network.

In the current scope of this specification, we assume that the UE implementing the residential SIP Telephony requirements interfaces with an analog phone.

This specification does not describe an implementation of the following features:

- Business telephone features
- Roaming or nomadic telephone features
- "Soft client" personal computer software "telephone" features

5.3 Functional Components for Residential SIP Telephony

The PacketCable architecture is based on IMS Release 6 architecture, so many elements will use the same name as in the IMS architecture. This architecture is more fully described in the PacketCable Architecture Framework Technical Report [ARCH-FRM TR]. Figure 1 illustrates the functional components of the architecture.

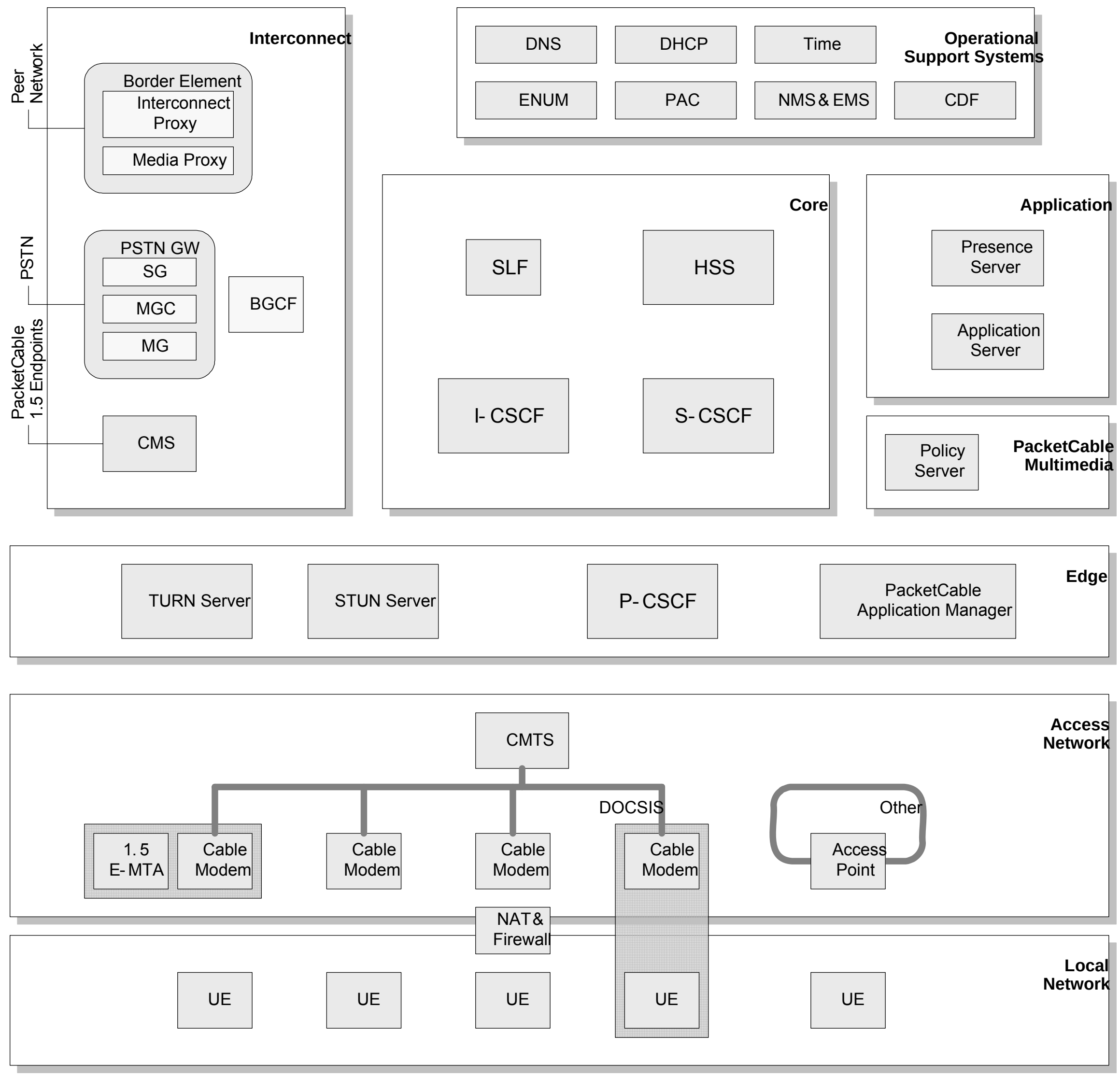

Figure 1 - PacketCable Functional Components for Residential SIP Telephony

For a description of the PacketCable Functional Components, please refer to [ARCH-FRM TR].

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].

An Application Server is a logical entity. In this specification, every feature that requires an Application Server is associated with a distinct one. For example, the Call Transfer section defines requirements on the Call Transfer Application Server (CT AS); the Outbound Call Blocking section defines requirements on the Outbound Call

Blocking Application Server (OCB AS) and so forth. In terms of implementation, more than one Application Servers could be combined into one physical entity.

5.4 Protocol Interfaces for Residential SIP Telephony

The protocol interfaces for Residential SIP Telephony are defined in [ARCH-FRM TR]. The RST feature specification (current document) is part of the RST suite of specifications which also consists of the following:

- 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. An embedded DVA is a DOCSIS cable modem (CM) integrated with a PacketCable DVA.

- PacketCable Residential SIP Telephony PACM Specification

  This document specifies the PACM requirements and the RST data model, as applicable to PacketCable Use providing PacketCable RST services.

6 NORMATIVE REFERENCE APPLICATION

The Residential SIP Telephony specification leverages a number of IETF documents and 3GPP, Telcordia, and CableLabs specifications. This section names and describes key specifications that contain requirements for implementations that claim compliance with this Residential SIP Telephony Feature Specification.

The PacketCable components compliant with the Residential SIP Telephony specification MUST implement all of the normative requirements defined within the specifications [PKT 23.228], [PKT 23.218], [PKT 24.229], [PAMI], [PKT 33.203], and [PKT 33.210], unless otherwise noted in the following sub-sections.

6.1 Adaptations of [PKT 24.229]

In order to support the PacketCable Residential SIP telephony application, several adaptations to [PKT 24.229] are required. In most cases, the adaptations involve strengthening of requirements; such as changing optional requirements to mandatory. The following requirements are modified. This is not a complete list, and there may be other adaptations specified throughout the document.

- A UE MUST include the P-Preferred-Identity header in SIP messages if it is assigned more than one Public Identity at registration time. [PKT 24.229] states that the P-Preferred-Identity header is optional regardless of the number of assigned public identities.
- Requirements on the use and population of Privacy and From headers are strengthened. For details, see Section 7.2, Caller ID.
- A UE MUST supply a Globally Routable User-Agent URI (GRUU) in its contact address when establishing a dialog. The requirements for the handling of a GRUU defined in [ID SIP-GRUU] MUST be supported by the UE.
- The requirements for the handling of a History-Info header defined in [RFC 4244] MUST be supported by the UE, Call Forwarding AS and SIP network elements. In addition to the procedures in [PKT 24.229], the UE MUST include a Supported Header in an INVITE request with the 'his info' option tag as defined in [RFC 4244]. When preying an INVITE, the SIP network elements MUST follow the rules defined in [RFC 4244] with regard to inclusion and population of the History-Info header.

6.2 Additional IETF Requirements Based on Specific IETF Rocs and Internet-Drafts

Additionally the requirements as defined within the following specifications MUST be adhered to in order to provide the service set defined herein.

- As defined in [RFC 3264], a UE MUST be capable of receiving SDP with a connection address of 0.0.0.0, in which case it means that neither RTP nor RTCP should be sent to the peer. This requirement provides backwards compatibility with [RFC 2543].
- The requirements within [ID SIP-ACR] MUST be supported by the ACR AS.
- The requirements for the handling of a Replaces header as defined by [RFC 3891] MUST be supported by the UE.
- The requirements for the handling of the REFER method as defined by [RFC 3515] MUST be supported by the UE.
- The requirements for the notification of message-waiting events as defined by [RFC 3842] MUST be supported by the Message Waiting Indicators (MWI) AS and the UE.

- The requirements defined by [RFC 4240] MUST be supported by an AS providing network announcements.
- The requirements in [RFC 3066] MUST be supported by a UE that wishes to indicate language preferences.
- The Alert-Info header as defined by [RFC 3261] MUST be supported by the Distinctive Ringing AS and the UE.
- Subscriptions to the dialog-event package as defined by [RFC 4235] MUST be supported by the UE.
- The requirements for the handling of the Join header as defined by [RFC 3911] MUST be supported by the UE and BLV AS.
- The requirements for the handling of a target-dialog header as defined by [RFC 4538] MUST be supported by the UE.
- The requirements for the handling of a P-Trace-Party-ID header as defined by [RFC 3603] MUST be supported by the UE and the COT AS.
- The UE MUST support the inclusion of a Presence Information Data Format Location Object (PIDF-LO) as defined by [RFC 4119] in INVITE requests conveying location information for emergency calls as required by [ID SIP-CONVEY].
- The DHCP option as defined by [RFC 3825] MUST be supported by the UE.
- The DHCP option as defined by [ID DHCP-CIVIC] MUST be supported by the UE.
- The emergency session URI defined by [ID SIP-URN] MUST be supported by the UE and the P-CSCF.
- The requirements defined by [RFC 3959] MUST be supported by the UE and any AS in the network that provides early media. For details, refer to Section 7.1.9.

7 RESIDENTIAL SIP TELEPHONY FEATURE REQUIREMENTS

7.1 Basic Calling Capabilities

All call flows in this specification are informative in nature. Some call flows illustrate the use of a 100 Trying response to a received INVITE implying the use of an unreliable transport protocol as opposed to reliable transport protocol as optionally provided by [PKT 33.203]. These are illustrative only and there is no normative requirement on the use of unreliable transport protocol imposed by this specification.

7.1.1 Digit Maps

Digit maps are provisioned on the UE to inform the UE about how dialed digits should be interpreted. A digit map is an ordered set of regular expressions combined with some special tokens that represent actions to be carried out by the UE when a regular expression is matched. While the remaining of this section is for North American digit maps, the PacketCable digit map syntax defined below can represent any country digit map.

PacketCable digit maps are capable of representing the following types of digit strings (where N is for descriptive purpose only and represents any single digit between 2 and 9 inclusive and X represents any single digit between 0 and 9 inclusive):

- NXX-XXXX (7-digit dialing)
- NXX-NXX-XXXX (10-digit dialing)
- 1-NXX-XXXX [terminator] (7-digit dialing)
- 1-NXX-NXX-XXXX (10-digit dialing)
- [0-1]N11 (special service codes)
- 0 [terminator] (local operator)
- 00 (carrier operator)
- 011 followed by up to 15 digits followed by [terminator] (international direct distance dialing)
- 01 followed by up to 15 digits followed by [terminator] (operator assisted international dialing)
- 10XXX [terminator] (3 digit carrier code dialing)
- 10XXX-NXX-NXX-XXXX (3-digit carrier code toll dialing)
- 101XXXX-NXX-NXX-XXXX (4-digit carrier code toll dialing)
- *XX (2-digit vertical service code)
- 11XX (2-digit vertical service code)
- *XXX (3-digit vertical service code)
- 11XXX (3-digit vertical service code)
- N [terminator] (1-digit speed dial code)
- NX [terminator] (2-digit speed dial code; first digit limited to 2-4)

Any time an optional terminator (represented above by "[terminator]") may be dialed, the UE MUST measure critical digit timing to determine if any additional digits are to be dialed. If the terminator digit is dialed (normally #), the UE MUST cancel critical digit timing measurement. Also, the UE SHOULD accept the digit string as dialed. If no terminator digit is dialed, the UE MUST accept the digit string as dialed when the critical digit timer expires.

If an operator requires the support of both 7- and 10-digit dialing, the 7-digit dialing should be specified with an optional terminator digit. This approach for defining the digit map is recommended because the first 3 digits dialed may not provide enough information to determine whether 4 or 7 additional digits are required to successfully match an entry in the digit map.

Dialing errors are captured through the digit map function. There are typically two kinds of dialing errors:

- The user dialed the wrong digits, or
- The user dialed an insufficient number of digits to create a complete address.

In order to capture errors where the user dialed the wrong digits, a digit map should contain a pattern designed to capture "all other dial strings" beneath all of the valid dial string pattern matches. When the "all other dial string" pattern is matched, an appropriate error response can be generated by the digit map function.

In order to capture errors where the user dialed an insufficient number of digits, the digit map contains the ability to specify timers. When a timer expires before a digit map pattern has been matched, the digit map matches the "timer expiration" pattern resulting in the appropriate action defined for the pattern. When no action is defined explicitly, the UE MUST perform the error operation typical for incomplete digit collection, such as play treatment or tone to the subscriber.

7.1.1.1 Digit Map Syntax Introduction

The digit map is organized into a list of rules. The UE MUST apply the digit map rules sequentially and upon matching a pattern, including timers, the UE MUST perform the specified action or actions.

All of the mechanisms in this syntax specification are described in both prose and an augmented Backus-Naur Form (BNF) defined in [RFC 4234]. [RFC 4234] defines a set of core rules that are used by this specification, and not repeated here. Implementers need to be familiar with the notation and content of [RFC 4234] in order to understand this specification. Certain basic rules are in uppercase, such as SP, LWS, HTAB, CRLF, DIGIT, ALPHA, etc.

Annex A of this document contains the augmented BNF for the Digit Map Syntax.

7.1.1.2 Digit Map Semantics

DigitMapPackage: This is a complete and self contained unit of definitions, suitable for provisioning on a UE. A UE MUST support a DigitMapPackage for each public user identity.

TimerDef : Defines the duration denoted by one of the named timers that influence the processing of dial strings. The duration is specified as a decimal number, in units of seconds. Fractional seconds may be specified. The UE MAY round the value up to the minimum required resolution to accommodate the resolution of its internal clocks. The minimum required resolution of the digit collection timers MUST be one tenth of one second. The values specified apply to all DigitMapPackages defined. Syntax checking, such as checking for redundant timer definitions, should be performed outside of the UE, prior to provisioning the digit map into the UE.

There are four named timers – T, S, L, and Z. Each timer has a different purpose:

- T - Start Timer: The length of time allowed to dial the first digit from the time dial tone is applied.
- S - Short Interdigit Timer: Used when critical timing should be performed, such as when the dialed digits constitute a complete address, but additional digits may constitute a different complete address.
- L - Long Interdigit Timer: Long interdigit timer. The allowable time between digits if the short interdigit timer has not been indicated in the digitmap.

- Z - Long Duration Timer: The duration a particular digit is to be held in order to be detected. When a Z is placed in a pattern, the Long Duration Timer is applied to the following key in the pattern.

SymbolDef: Defines a named symbol that may be referenced within a pattern or action. The value of the symbol is a string formed by concatenating all of the constant pieces in the definition, but omitting the surrounding quotation marks and any intervening white space and comments. SymbolDefs apply to the entire DigitMapPackage, and each SymbolDefs name must be unique. When referencing symbols, the name comparison must be case insensitive. The method of specifying the value of a symbol is similar to the method used to specify the value of parameter in a rule action.

MapDef: Defines a named map. MapDefs apply to the entire DigitMapPackage, and it is an error to define the same name more than once, as a symbol or a map. A map contains one or more rules for interpreting dialed digits. All the rules in a map are compared to the same dialstring in parallel while dialing is in progress, and continues until at least one of the rules has matched, or until it is determined that none of the rules can match the dialstring. If more than one rule matches the same dialstring, then the first matching rule is followed.

Rule: A rule includes a pattern that can be matched against a dial string, and a list of actions to be carried out, in sequence, when a dialed string is matched.

Pattern: A pattern is a concise way of representing a number of dial strings of interest. In its simplest form, a pattern denotes a sequence of keys that are pressed on a telephone keypad. The valid keys are the numbers '0' through '9', the '*' key, the '#' key, and the keys 'A' through 'D' that are present on some keypads. Additional elements may be used so that a single pattern may match more than one dial string.

The character 'X' (or 'x') denotes any numerical digit – '0' through '9'.

The notation '[stu…z]', where 's', 't', etc. are letters on keyboard keys (in case the UE has the ability to type letters on a keyboard), matches any of the keys 's', 't', etc. The order of the characters within the brackets is unimportant.

The notation '[n-m]', where 'n' and 'm' are digits, matches any single digit in the range n-m. In other words, '[3-7]' means the same as '[34567]'. This form may also be combined with the preceding form, so that '[A2-49]' is valid and is equivalent to '[2349A]'

The notation '[^stu…z]' matches any keyboard key that would not be matched by '[stu…z]'. This may be used with all the forms above, so that '[^A2-49]' is valid and is equivalent to '[015678BCD*#]'

Any key or bracketed notation for a key may be preceded by the character 'Z' (or 'z'). This means to apply the long duration timer Z to that keypress. In that case, a match occurs only if a matching key is held down for a duration equal to the long duration timer Z.

When the notation for a key is followed by '{n-m}' it means that the preceding key should be repeated a minimum of 'n' times and a maximum of 'm' times. This is equivalent to creating multiple patterns. For instance "1x{2-3}7" is a matching equivalent to the two patterns "1xx7" and "1xxx7." The form '{-m}' is allowed and means '{0-m}'.

The characters 'T', 'S', and 'L:' (or 't', 's', and 'l') may be included to denote the corresponding timer. These elements are matched when the corresponding amount of time elapses without key entry since the entry of the previous matched key.

Anything valid in a pattern, including all of the above forms, may be enclosed in parenthesis '(' and ')' and then inserted as an element in a pattern, denoting a sub-pattern. Sub-patterns may be nested within other sub-patterns. Within a pattern, sub-patterns may be denoted by counting the total number of '(' characters from the beginning of the pattern to the beginning of the sub-pattern. For example, in the pattern "1(8xx)(555(xxxx))" the sub-pattern 'xxxx' may be referenced by the value 3. This is useful in constructing the parameters to actions.

A sub-pattern may also contain an '=' followed by the name of a symbol or map. If a symbol is named, then the meaning is the same as if the string value of the symbol had been included. In this case the value of the symbol must conform to the syntax of a pattern. If a map is named, then that map is applied, and then matching continues with whatever follows the sub-pattern.

Note: The execution of an action associated with a sub-pattern is conditional on a successful match of the parent pattern. That means that the action specified by the sub-pattern cannot be applied until successful match of the entire pattern.

Finally, the characters '-', '.' and space that are often used as punctuation in telephone numbers may be used freely anywhere a character denoting a key may be used to improve the readability of patterns. The '-', '.' and space characters must be ignored as they play no role in matching.

Action: An Action invokes specific behavior when a rule has been matched by a dial string. Two special actions (RETURN and USEMAP) affect digit map processing and are defined here. Other actions may be defined to interact with the environment outside of the digit map, and are defined elsewhere.

Each action has a name, and may have one or more parameters. If the action has parameters, they are enclosed in parentheses and separated by commas. There are two kinds of parameters: map parameters and string parameters. Each action has some expectation regarding the number of parameters it expects, which ones are maps, which are strings, and what string content is acceptable.

A map parameter is used to pass a map reference to an action. Its most obvious use is with the USEMAP action. A map parameter is denoted by an '=' followed by the name of a map.

A string parameter is composed of one or more strings. Each string is concatenated to form the value of the parameter. (The form used to denote a parameter is similar to that used to denote the value of a SymbolDef.) Each string making up the parameter is demarked using single or double quote characters. A string may contain any printable ascii character except single or double quotes.

A string in a string parameter may also refer to characters matched in the pattern of the containing rule. The string beginning with '#' followed by a number N, refers to the portion of the dialstring matching the Nth sub-pattern within the pattern. For example, given the pattern "1(8xx)(555(xxxx))," and the dialstring "18885559876", "#3" refers to the value "9876." The string "#0" refers to the entire matched dialstring.

A string may also be denoted by a '#' followed by a number N. If sub-pattern N is a reference to a digit map, then this syntax refers to the value returned by that map (see description of RETURN). Action RETURN: The RETURN action specifies the value that results from the application of a map. It may have zero or one parameter. The value returned for zero parameters is "" (empty string). The value retuned for one parameter is the digits matched from the dialstring.

The RETURN action only affects the value of the map. It does not affect other actions associated with the same rule. (Hence the RETURN action MAY be followed by other actions, though this may be confusing.) There SHOULD be at most one RETURN action per rule. For completeness, if there is more than one, the last one determines the value of the map. If there are no RETURN actions in a rule, then a RETURN action with no parameters is assumed.

Action USEMAP: The USEMAP action is a way to apply a named map to dialstring input received after the containing rule has been matched. It provides a way for dialed digits to determine how subsequent dialed digits should be interpreted. It may have no parameters or one map parameter. If a parameter is present, it denotes the map to be used. If no parameters are present, then USEMAP processes the first map defined within the DigitMapPackage.

The referenced map may also have rules that specify actions. Actions are processed in order within a rule. When a USEMAP action is processed, any actions it may specify are processed after actions preceding the USEMAP, and before actions that follow the USEMAP.

There is a subtlety here that interacts with that for sub-pattern matching. As noted above, if a sub-pattern references a map, and if the reference to the sub-pattern is followed by other targets to be matched, then the rules for the sub-pattern cannot be invoked until the parent pattern is known to match. If the sub-pattern contains a USEMAP, then at the time the USEMAP is applied, the dial string input will have been advanced beyond where it was when the rule containing the USEMAP was matched.

Action MAKE-CALL: The MAKE-CALL action provides a standard way to create a SIP or TEL URI and send an INVITE to the target identity. The MAKE-CALL action's parameter is the Request-URI to be used to create the INVITE request.

7.1.1.3 Sample Digit Map

```
// Timer values
TIMER T=16    // Start timer. The length of time allowed to dial the first digit
              // from the time dial tone is applied
TIMER S=4     // Short interdigit timer. Used when critical timing should be
              // performed, such as when the dialed digits constitute a complete
              // address, but additional digits may constitute a different
              // complete address
TIMER L=16    // Long interdigit timer. The allowable time between digits if the
              // short interdigit timer has not been indicated in the digitmap.
TIMER Z=2.0   // Long duration timer. The duration a particular digit is to be
              // held in order to be detected.

// Symbols
domain = "@example.net"
areaCode = "303"
dialString = ";user=dialstring"   // Just to shorten things
homeEmergencyNumber = "911"
localEmergencyNumber = "911"      // alternate emergency number

// Maps
MAP MainTable = // This is where processing starts
       "T"                 : REORDER    // Reorder Tone or Annc.
       "0S"                : MAKE-CALL ("sip:0" =domain =dialstring)
       "0#"                : MAKE-CALL ("sip:0" =domain =dialstring)      "00"
       : MAKE-CALL ("sip:00" =domain =dialstring)
       "(=Emergency)"      : EMERGENCY-CALL ("urn:service:sos")
       "[2-8]11"           : MAKE-CALL ("sip:" #0 =domain =dialstring)
                           // or  : MAKE-CALL ("tel:" #0 ";phone-context=+1")
       "(=PhoneNumbers)"   : MAKE-CALL ("tel:" #1v )
       "(=DialAround)"     : MAKE-CALL ( "tel:" #1v )
       "0(=PhoneNumbers)"  : MAKE-CALL ( "sip:0" #1 =domain =dialstring )
       "0([2-9]x{0-15})S"  // let operator decipher unknown string
                           : MAKE-CALL ( "sip:0" #1 =domain =dialstring )
"([2-9])S"          : MAKE-CALL ( "sip:" #1 =domain =dialstring)
                                  // one-digit speed dialing
       "([2-9])#"          : MAKE-CALL ( "sip:" #1 =domain =dialstring)
                                  // one-digit speed dialing
       "([2-4])xS"         : MAKE-CALL ( "sip:" #1 =domain =dialstring)
                                  // two-digit speed dialing
       "([2-4])x#"         : MAKE-CALL ( "sip:" #1 =domain =dialstring)
                                  // two digit speed dialing
       "(=ImmediateVSCs)"  : RETURN
        "(=DelayedVSCs)"   : RETURN
       "Z#"                : RECALL; USEMAP(=MainTable)
                                  // Press # for 2 seconds and get recall dial tone
       "(x{1-15})"         : REORDER    // Any other digit string matches here
```

```
MAP Emergency =     // Matches emergency dialstrings
      "(=localEmergencyNumber)"        : RETURN
      "(=homeEmergencyNumber)"         : RETURN
      "[01](=homeEmergencyNumber)"     : RETURN

MAP PhoneNumbers =  // Matches phone# and returns the canonicalized
                    // form without a scheme
      "(=CollectNumbers)"       : RETURN( #1v ) // override later matches
      "(=SurchargedNumbers)"    : RETURN( #1v ) // override later matches
      "([2-9]x{6})"             : RETURN( "+1" =areaCode #1 )
      "([2-9]x{2}[2-9]x{6})"    : RETURN( "+1" #1 )
                                                // This is ambiguous with above
      "1([2-9]x{6})"            : RETURN( "+1" =areaCode #1 )
      "1([2-9]x{2}[2-9]x{6})"   : RETURN( "+1" #1 )
                                                // This is ambiguous with above
      "011([2-9]x{1-14})"       : RETURN( "+" #1 )
      "011([2-9]x{1-14})#"      : RETURN( "+" #1 )
      "01([2-9]x{1-14})"        : RETURN( "+" #1 )
      "01([2-9]x{1-14})#"       : RETURN( "+" #1 )

MAP CollectNumbers = // 800 and friends
      "(800[2-9]x{6})"          : RETURN( "+1" #1 )
      "(866[2-9]x{6})"          : RETURN( "+1" #1 )
      "(888[2-9]x{6})"          : RETURN( "+1" #1 )
      "1(800[2-9]x{6})"         : RETURN( "+1" #1 )
      "1(866[2-9]x{6})"         : RETURN( "+1" #1 )
      "1(888[2-9]x{6})"         : RETURN( "+1" #1 )

MAP SurchargedNumbers = // 900, etc.
      "(900[2-9]x{6})"          : RETURN( "+1" #1 )
      "1(900[2-9]x{6})"         : RETURN( "+1" #1 )


MAP DialAround =    // Matches dial around phone# and returns canonicalized
                    // form without scheme
      "10([2-9]x{2})(=PhoneNumbers)"
                                : RETURN( #2v ",cic=+1" #1 )
                                 // (3 digit CIC dialed by user)
      "101(x{4})(=PhoneNumbers)"
                                : RETURN( #2v ",cic=+1" #1 )
                                 // (4 digit CIC dialed by user)

MAP ImmediateVSCs = // Matches and executes immediate VSCs. Returns nothing
      "*57"  // (COT-ACTIVATE)
                                : COT-ACTIVATE      // Can't use MAKE-CALL for this.
      "*63"  // (SCF)
                                : MAKE-CALL ( "sip:" #0" =domain =dialstring)
      "*66"  // (AC-ACTIVATE)
                                : AC-ACTIVATE // Auto Callback
      "*69"  // (AR-ACTIVATE)
                                : AR-ACTIVATE // Auto Recall
      "*74"  // (SD8)
                                : RECALL; USEMAP (=SD8)
      "*75"  // (SD30)
                                : RECALL; USEMAP (=SD30)
      "*[78]7"      // (ACR-ACTIVATE/ACR-DEACTIVATE)
                                : MAKE-CALL ( "sip:" #0 =domain = dialstring)
      "*83"  // (SCF)
                                : SCF-PROGRAM ( "sip:" #0 =domain = dialstring)
      "*9[01]"      // (DND-ACTIVATE/DND-DEACTIVATE)
                                : MAKE-CALL ( "sip:" #0 =domain = dialstring)
      "*95"  // (SPP)
                                : SPP-PROGRAM ( "sip:" #0 =domain = dialstring)
      "*96"   // (SB-MAINT)
                                : SB-MAINT  // solicitor blocking maintenance
      "*72"  // (CFV-ACTIVATE)
```

```
                    //    Play Recall Dial Tone
                    //    Collect digits using subset digitmap
                    //    Prepend "*72." onto dialed address using SIP URI
                    //    The following is an example implementation:

                                  : RECALL; USEMAP(ForwardingNumber)

       "*73"  // (CFV-DEACTIVATE) (reuse *72 for this)
                                  : MAKE-CALL ( "sip:*72." =domain =dialstring)

MAP DelayedVSCs = // Make some state change, then continue processing dialing
       "*92"  // (HOLD)
                                  : HOLD-ACTIVATE; RECALL; USEMAP(=MainTable)
       "*67"  // CIDS-SUPRESS
                                  : CID-SUPPRESS
       "*82"  // CIDS-DELIVER
                                  : CID-DELIVER
       "*70"  // (CCS) (Really Toggle-Call-Waiting)
                                  : CALL-WAITING("toggle"); RECALL; USEMAP(=MainTable)
              // The CALL-WAITING action must give the right tone.

MAP SD8 =     // Program one-digit speed dial number
       "[2-9]"                    : SD-PROGRAM ( "sip:*74." #0 =domain
                                                 =dialstring)
       "[^2-9]"                   : REORDER
       "s"                        : REORDER

MAP SD30 =    // Program two-digit speed dial number
       "[2-4]x"                   : SD-PROGRAM ( "sip:*75." #0 =domain
                                                   =dialstring)
       "[^2-4]"                   : REORDER
       "s"                        : REORDER
       "[2-4]s"                   : REORDER

MAP ForwardingNumber = // Just for programming CFV
       "(=SurchargedNumbers)"     : REORDER // not allowed
       "(=PhoneNumbers)"          : MAKE-CALL ( "sip:*72." #1 =domain
                                                 =dialstring)
```

7.1.2 Vertical Service Codes

Vertical service codes are dialed digit strings used to represent feature invocations. They are sometimes referred to as star codes or feature activation codes. A feature invocation is feature activation, feature deactivation, feature programming, or feature instance. Vertical service codes (VSCs) do not have to begin with a star, nor are they restricted to 3-digit strings such as "*69." The actual digits that make up a VSC are defined by the digit map. The digit map also defines whether the VSC is processed by the UE or forwarded to the network via the substitution methodology defined above.

Since VSCs can be defined by an operator based on their needs and some VSCs are processed by the UE, the digitmap indicates how to map VSCs to PacketCable residential SIP services.

For vertical service codes processed in the network, the UE MUST pass the VSC and any other digits required by the digit map to the network using a SIP INVITE. The digits MUST be reported using a SIP URI with user=dialstring. The digits themselves are included in the SIP URI as defined by the digitmap substitution.

Table 1 provides the defined local action identifiers for the services defined in this document, as well as their suggested VSCs for use in North America.

Table 1 - Digit Map Actions

Service	Action	UE Action ID	Suggested VSC
Anonymous Call Rejection	Activation	ACR-ACTIVATE	*77
	Deactivation	ACR-DEACTIVATE	*87
Automatic Callback	Activation	AC-ACTIVATE	*66
Basic Call	Activation	MAKE-CALL	none
Automatic Recall	Activation	AR-ACTIVATE	*69
Caller ID Per-Call Blocking	Suppress	CID-SUPPRESS	*67
Caller ID Per-Call Delivery	Deliver	CID-DELIVER	*82
Customer Originated Trace	Activation	COT-ACTIVATE	*57
Call Waiting	Toggle	CW-TOGGLE	*70
Call Forward Variable	Programming	CFV-PROGRAM	*72
	Deactivation	CFV-DEACTIVATE	*73
Distinctive Alerting	List Maintenance	DA-MAINT	*61, *81
Do Not Disturb	Programming	DND-PROGRAM	*78, *79
Hold	Activation	HOLD-ACTIVATE	*52
Selective Call Forward	Programming	SCF-PROGRAM	*63, *83
Solicitor Blocking	List Maintenance	SB-MAINT	Two unique codes
Speed Dial	Programming	SD-PROGRAM	*74 (SD8), *75 (SD30)
Subscriber Programmable PIN	Programming	SPP-PROGRAM	Any unique code
Emergency Call	Activation	EMERGENCY-CALL	911, 0911, 1911 (or other provisioned emergency number)

7.1.3 Addressing and Dial String Entry

A UE MUST be able to collect digits, aggregate digits into dial strings and match them with the digit patterns specified in a provisionable digit map. Dial strings are strings of characters that correspond to DTMF digits. The characters are 0-9, A-D, *, and #.

The form of an address created by a UE is determined by the provisioned digit map. For encoding of feature codes with the "*" character in them, the digit map MUST create a SIP URI with a "user=dialstring" parameter, as in the following example.

sip:*72@example.com; user=dialstring

The last rule of the digit map specifies the default behavior, and MUST match any pattern of dialed digits. This last rule of the digit map MUST provide an error indication to the user that an invalid pattern of dialed digits has been entered. The last rule of the digit map SHOULD specify reorder tone be played to the user.

Once a dial string matches a digit pattern, the UE MUST stop digit collection.

The digit map MUST be preconfigured on the UE via PACM.

A UE MUST take action according to the provisioned digit map processing after successfully matching the dialstring. If the output of the digit map is intended to invoke a feature, the feature is performed by the UE, or another network element, or a combination of both.

7.1.4 Basic Call Signaling

The basic calling capabilities consist of an originating call half and a terminating call half. A UE MUST support [PKT 24.229] for the basic call capabilities. A UE MUST support [RFC 4566]. The requirements specified in this section apply in the absence of other features. The UE MUST support the tones specified in Table 2. The values provided in Table 2 are only recommended values and may vary based on the country of operation.

Table 2 - North American Supervision and Call Progress Tones

Meaning	Frequencies Hz	Signal Level (see Note 1 below)	Cadence
Dial Tone	350 + 440	-10dBm0, -13dBm0 per freq.	
Recall Dial Tone (also called Special Conditions Dial Tone)	350 + 440	-10dBm0, -13dBm0 per freq.	3 bursts (100ms On, 100ms Off), steady On
Message Waiting	350 + 440	-10dBm0, -13dBm0 per freq.	10 bursts (100ms On, 100ms Off), steady On
Audible ringing	440+480	-16dBm0, -19dBm0 per freq.	Continuous Repeat (2s On, 4s Off)
Busy	480 + 620	-21dBm0, -24dBm0 per freq.	Continuous Repeat (500ms On, 500ms Off)
Reorder	480 + 620	-21dBm0, -24dBm0 per freq.	Continuous Repeat (250ms On, 250ms Off)
Receiver Off Hook (Howler)	1400 + 2060 + 2450 + 2600	-6 to +3 dBm0	Objective is to transmit highest power available, Continuous Repeat (100ms On, 100ms Off)
Call Waiting 1	440	-13dBm0	Call Waiting Cadence, 300ms On
Call Waiting 2	440	-13dBm0	100ms On, 100ms Off, 100ms On
Call Waiting 3	440	-13dBm0	100ms On, 100ms Off, 100ms On, 100ms Off, 100ms On
Call Waiting 4	440	-13dBm0	100ms On, 100ms Off, 300ms On, 100ms Off, 100ms On
High Tone	480	-17dBm0, -24dBm0 per freq.	
Low Tone	480 + 620	-21dBm0, -24dBm0 per freq.	
Confirmation	350 + 440	-10dBm0, -13dBm0 per freq.,	100ms On, 100ms Off, 100ms On, 100ms Off, 100ms On
Error	620	-13dBm0	
Note 1: dBm0 is the network reference PCM code level not including the analog loss plan			

7.1.4.1 Originating a Call

When originating a call, a UE MUST be able to:

- Detect off hook.
- Apply dial tone when off hook is detected or when specified by the provisioned digit map.
- Collect dialed digits as specified in the provisioned digit map.
- Process the actions specified in the provisioned digit map.
- Initiate a SIP session, as specified in [PKT 24.229].

- Encode and decode RTP and RTCP packets as defined in [CODEC-MEDIA].

When a UE sends an INVITE to initiate a session, the UE MUST populate the offer SDP as per the feature data SDP Parameters and value defined in Table 4.

When a UE sends an INVITE to initiate a session, the UE MUST record the identity of the called party in local memory. If the identity of the called party is anonymous or unknown, the UE MUST record the information that the called party was anonymous or unknown in local memory. The identity of the called party may be unknown in the case of the Auto Callback feature.

Upon receiving an 18X message with an alert-info header, the UE MUST apply the ringback tone identified in the alert-info header to the phone if it has the capability to do so. If the UE is not capable of applying the ringback tone identified in the alert-info header, it MUST locally apply ringback tone. If the UE receives a 180 or 183 provisional response message without an alert-info header and without an SDP, the UE MUST locally apply ringback tone to the phone, and any received RTP packets should be ignored. Upon receiving a 180 or 183 message without an alert-info header and with an SDP, and if RTP audio is received, the UE MUST play back the received RTP audio rather than applying local ringback tone to the phone. Upon receiving a 180 or 183 message without an alert-info header and with an SDP, and there is no RTP audio received, then the UE MUST locally apply ringback tone. These actions are detailed in Table 3.

Table 3 - Actions at Calling UE upon Receipt of an 18X Message

Alert-Info Header Present?	SDP Received?	RTP Packets Received?	Action
No	No	No	Play local ringback
No	No	Yes	Play local ringback
No	Yes	No	Play local ringback
No	Yes	Yes	Play audio received in RTP packets
Yes	No	No	Play audio from alert-info header
Yes	No	Yes	Play audio from alert-info header
Yes	Yes	No	Play audio from alert-info header
Yes	Yes	Yes	Play audio from alert-info header

Upon receiving a 486 BUSY HERE message, the UE MUST apply an indication to the user of the call status as defined by the entry for this response code in the announcement map as specified in Section 7.1.6.6, respond with an ACK and terminate the session.

Upon receiving any other response code, the UE MUST respond according to Section 7.1.11.

7.1.4.2 Receiving a Call

When receiving a call, a UE MUST be able to:

- Play ringback to the calling party
- Locally apply ringing or distinctive ringing to the called party
- Encode and decode RTP packets

A UE MUST reject received RTP packets that are not associated with an existing early or confirmed dialog. Refer to [RFC 3261] for the definition of early and confirmed dialogs.

A UE MUST have the capability to accept an incoming SIP session, as specified in [PKT 24.229]. For SDP negotiation, the UE MUST use the SDP parameters and value defined in Table 4.

When a UE receives an incoming INVITE, and sends a progress message according to [PKT 24.229], it MUST alert the user.

When a UE receives an incoming INVITE, and if the UE has alerted the user, the UE MUST record the dialog ID of the call and the identity of the calling party, if known, in local memory. If the identity of the calling party is anonymous or unknown, and if the UE has alerted the user, the UE MUST record the information that the calling party was anonymous or unknown in local memory.

The UE MUST respond with a 486 BUSY HERE message if the phone is already off-hook when the UE receives the incoming INVITE unless Call Waiting is enabled for this user as defined in Section 7.5.2. The UE MUST NOT record the identity of the calling party in its incoming call memory if the UE responds with a 486 BUSY HERE message.

After the user has been alerted about an incoming call and the UE detects an off hook event, the UE MUST return a 200 OK to the INVITE.

7.1.4.3 On Hook Processing

The following requirements apply when a UE with an active session (where the called party has answered the call and two-way media is being exchanged) detects an on-hook condition before receiving a BYE from the remote party.

- The UE MUST start the BYE delay timer which delays the sending of BYE to allow the user to pick up the phone again (as defined in Table 4, 'BYE Delay') and send comfort noise RTP packets to the remote UE using the RTP payload format defined in [RFC 3389]. While the BYE delay timer is running, the previously established session is still active, hence feature interactions are maintained.
- If the UE detects off hook before expiry of BYE Delay timer, the UE MUST stop the BYE delay timer and stop sending comfort noise RTP packets to the remote UE according to [RFC 3389] and resume the media session according to the previously negotiated SDP modes.
- If the BYE Delay timer expires, the UE MUST send a BYE request.
- If a BYE request is received while the BYE Delay timer is running, the UE MUST stop the BYE Delay timer and respond to the BYE with a 200 OK.

7.1.4.4 Extended Off Hook Processing

There are two ways a telephone handset can be left off-hook for an extended period of time:

- Origination Mode: the device was in an idle state (on-hook) and is placed off-hook and left there (presumably accidentally) for a long period of time, or
- Termination Mode: the device was in use for a call and was not successfully returned to an idle state (placed on hook) within 20 seconds after the call was terminated (that is, this UE received BYE). 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 prompt the end user to dial the desired number according to Section 7.1.7. After the Origination Mode Dial Tone timer defined in Table 4 expires, before which the user could have committed a dialing error or simply did not dial any digit, the UE MUST invoke its permanent sequence and lockout treatment as defined below.

In the Termination Mode, the UE MUST apply the Termination Mode Off-Hook Error Signal (defined in Table 4) for a time period defined as Termination Mode Error Signal Timer in Table 4 or until an on-hook state is detected. If the Termination Mode Error Signal Timer expires before the on-hook state is detected, the UE MUST play the announcements as defined in Table 4 and then invoke the permanent sequence and lockout treatment as defined below. Upon detecting the on hook state, the UE MUST return to the idle state for the provisioned services.

When the permanent sequence and lockout treatment is invoked, the UE MUST begin to generate a configurable sequence of tones and announcements. These tones (Permanent Sequence Tone 1, Permanent Sequence Tone 2 and Permanent Sequence Tone 3) and the timers that govern the duration of these tones (Permanent Sequence Timer 1, Permanent Sequence Timer 2 and Permanent Sequence Timer 3) are defined in Table 4. 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 which increases in amplitude and repeats for a provisioned period of time. This tone can be re-defined by provisioning a different tone.

The sequence of tones and announcements MUST be generated by the UE until the last timer in the sequence 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 (the Lockout Reset Timer defined in Table 4). After this Lockout Reset Timer expires, the analog line MUST return to normal operation, responding to the next originator mode off hook state according to Section 7.1.7, or in termination mode with appropriate (ring, caller ID, etc.) signaling.

7.1.4.5 Feature Data

Table 4 summarizes the feature data elements defined to support implementation of the basic call feature.

Table 4 - Basic Call Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Digit map	Digit Map	Volatile in UE, Non-volatile in XDS	Per public user ID	XDS	XDS	UE	Mandatory
SDP parameters and value (Note 1)	Character string	Volatile in UE, Non-volatile in XDS	Per public user ID	XDS	XDS	UE	Mandatory
BYE Delay	Integer (seconds)	Volatile in UE, Non-volatile in XDS	Per network	XDS	XDS	UE	Mandatory
Origination Mode Dial Tone Timer	Integer (seconds)	Volatile in UE, Non-volatile in XDS	Per network	XDS	XDS	UE	Mandatory
Termination Mode Off-Hook Error Signal	Tone	Volatile in UE, Non-volatile in XDS	Per network	XDS	XDS	UE	Mandatory
Termination Mode Error Signal Timer	Integer (seconds)	Volatile in UE, Non-volatile in XDS	Per network	XDS	XDS	UE	Mandatory
Permanent Sequence Tone 1	Tone	Volatile in UE, Non-volatile in XDS	Per network	XDS	XDS	UE	Mandatory
Permanent Sequence Timer 1	Integer (seconds)	Volatile in UE, Non-volatile in XDS	Per network	XDS	XDS	UE	Mandatory
Permanent Sequence Tone 2	Tone	Volatile in UE, Non-volatile in XDS	Per network	XDS	XDS	UE	Mandatory
Permanent Sequence Timer 2	Integer (seconds)	Volatile in UE, Non-volatile in XDS	Per network	XDS	XDS	UE	Mandatory
Permanent Sequence Tone 3	Tone	Volatile in UE, Non-volatile in XDS	Per network	XDS	XDS	UE	Mandatory

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Permanent Sequence Timer 3	Integer (seconds)	Volatile in UE, Non-volatile in XDS	Per network	XDS	XDS	UE	Mandatory
Lockout Reset Timer	Integer (seconds)	Volatile in UE, Non-volatile in XDS	Per network	XDS	XDS	UE	Mandatory
Note 1: The SDP parameters are described in the SDP profile included in [PKT 24.229]. Parameters that are bound to exclusive resources, such as port numbers, should be left to the UE.							

7.1.4.6 Feature Interactions

Please see each feature section for feature interactions with basic calls.

7.1.4.7 Call Flows

The call flows in this section illustrate basic call scenarios. The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

Figure 2 illustrates a call from one UE to another UE where the call is answered.

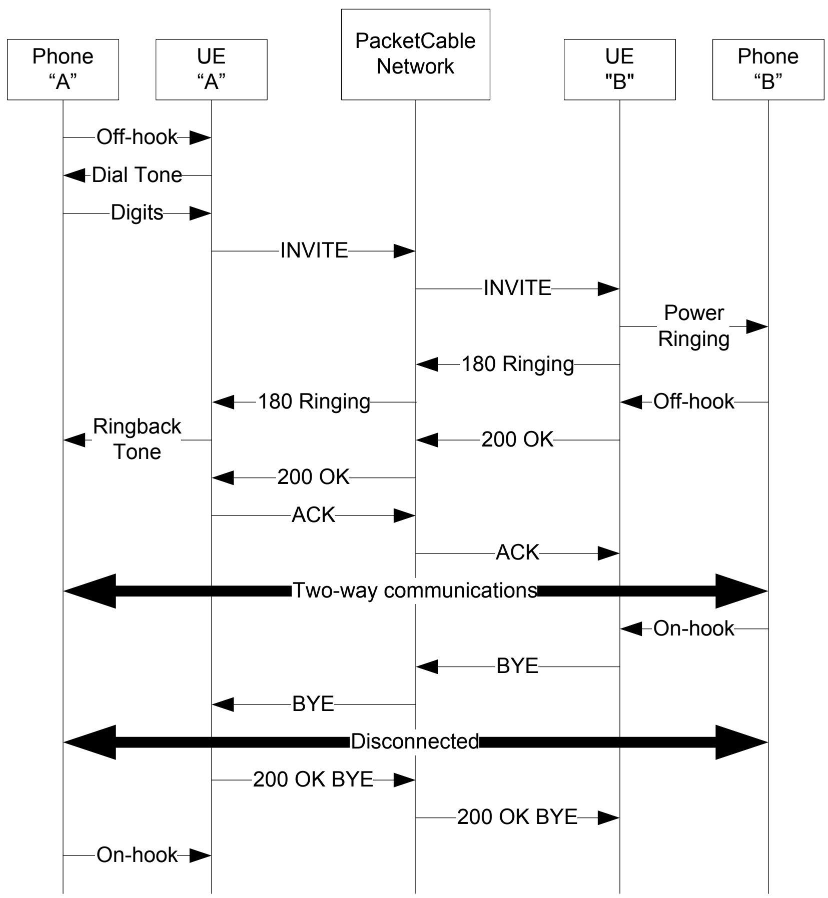

Figure 2 - Basic Call; On-Net to On-Net

Figure 3 illustrates a call from one UE to another UE where the call is not answered. A provisionable timer (defined in Table 11) is used on the called UE to decide when to stop ringing and send back an error response to the caller.

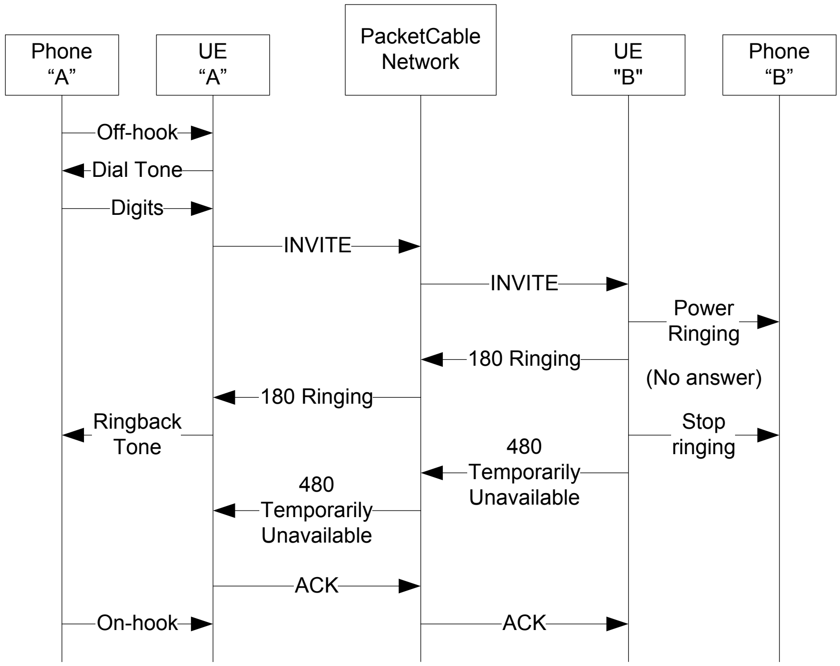

Figure 3 - Basic Call; On-Net to On-Net No Answer

Figure 4 illustrates a call from one UE to another UE where the called UE is busy (on an active call), assuming that call waiting is not applied at the called UE.

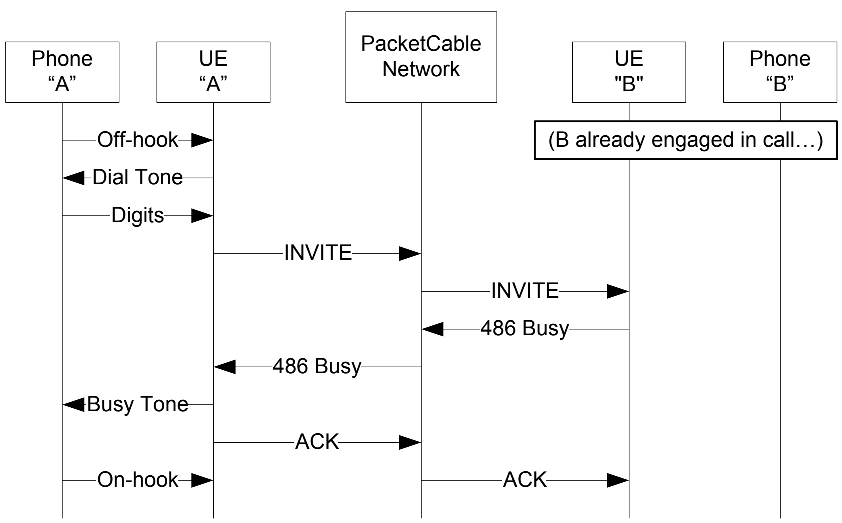

Figure 4 - Basic Call; On-Net to On-Net Busy, Call Waiting Not Applied

Figure 5 illustrates a call from a UE to a phone on the PSTN network.

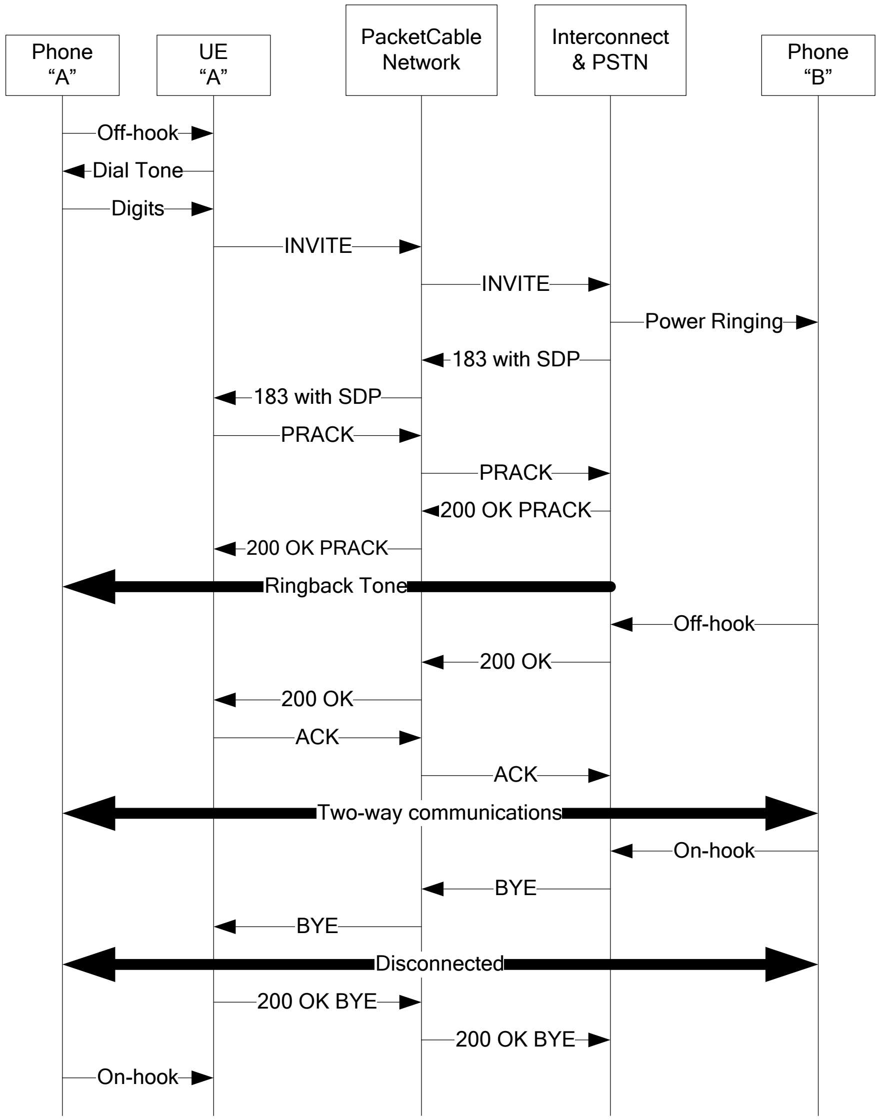

Figure 5 - Basic Call; On-Net to PSTN Off-Net

Figure 6 illustrates a call from a phone on the PSTN network to a UE (on the PacketCable network).

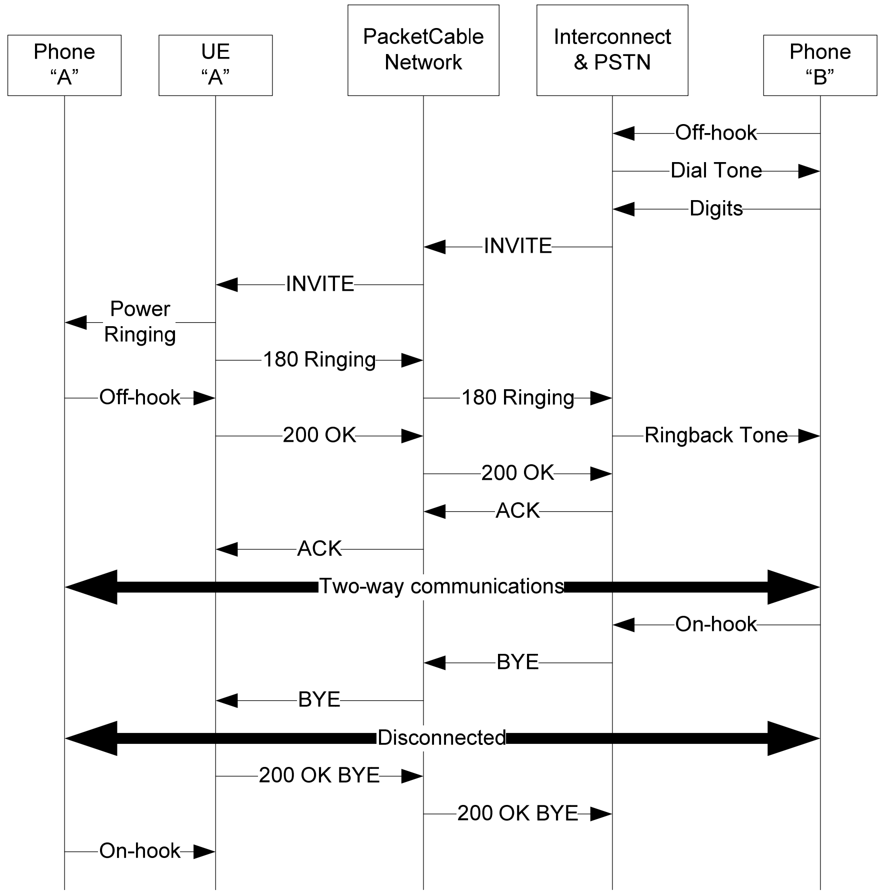

Figure 6 - Basic Call - PSTN Off-Net to On-Net

7.1.5 Held Media

There are a number of service capabilities defined in this specification that require an active bi-directional media stream to be held.

For the purposes of this specification a held media stream is defined to be a negotiated (via the SIP SDP offer-answer procedures as defined in [RFC 3264]) unicast media stream between two parties where the offer-answer exchange between the parties results in a negotiated directionality attribute of inactive. Where one of the parties is a UE not conformant to this specification, the media attributes may be negotiated to a value other than inactive; this is discussed in Section 7.1.5.1.

This specification assumes that there is only a single negotiated audio media stream between the two parties; support for multiple negotiated active media streams (that is, additional audio, video, image etc) is out of the scope of this specification.

7.1.5.1 Feature Execution

The UE MUST implement all the SDP requirements defined in [RFC 4566], including all the mandatory requirements and all the recommended ("should") requirements unless otherwise noted in this section.

While the requirements detailed are written with reference to the interaction between UEs, they MUST apply equally to any PacketCable components between which an INVITE initiated SIP dialog can be established.

In the procedures defined within this section reference is made to the 'controlling UE' and the 'held UE'. The 'controlling UE' is the party in the call that initiates the hold action; the 'held UE' is the party in the call being held by the 'controlling UE'.

A dual-hold state is where both parties have each placed the other 'on hold'.

7.1.5.1.1 Actions at the UE initiating hold

A controlling UE what wants to place a media stream "on hold" MUST offer updated SDP to the other media endpoint with an attribute of a=inactive in the media description block.

A controlling UE that wants to place a media stream "on hold" MUST NOT set the connection information of the SDP to a null IP address (for example, it MUST NOT set the 'c=' connection line to c=IN IP4 0.0.0.0).

A controlling UE that wants to place a media stream "on hold" SHOULD locally mute the media stream.

When the 'held UE' wishes to place the other party on hold, the UE MUST adhere to the requirements detailed within this section.

7.1.5.1.2 Actions at the UE initiating hold on receiving a hold 'answer'

In response to offering a "hold" SDP, a controlling UE will receive a corresponding answering SDP. If this answering SDP contains an a=inactive attribute in either the media block equating to that being held or the session level, then the hold action has been successfully completed.

If no directionality (these being sendrecv, sendonly, recvonly, inactive) attribute is received either in the answering SDP's media block or the session level of the SDP; or a directionality attribute other than a=inactive is received, then the recipient of the hold offer SDP is not conformant to either this specification or [RFC 3264]. If the recipient of the hold offer SDP is not conformant to either this specification or [RFC 3264] as indicated by the answering SDP, then the controlling UE MUST offer an updated SDP to the other party that contains a connection data line in the media block of the SDP set to c=0.0.0.0 (as required by [RFC 2543]). Reception of an answering SDP means that the hold action has been successfully completed.

7.1.5.1.3 Actions at the UE receiving the hold offer

When a UE receives an offering SDP with an attribute of a=inactive in the media block (placing the media stream "on hold"), then the UE MUST answer with an updated SDP and include an SDP media attribute of a=inactive in the media description of the answer SDP equating to the offered held media.

When a UE receives an offering SDP with an attribute of a=inactive in the media block (placing the media stream "on hold"), then the UE MUST NOT set the connection data of the SDP to c=0.0.0.0.

A held UE SHOULD provide comfort noise locally to the subscriber. Comfort noise is artificial background noise used to fill the silence. The exact definition of comfort noise is out of scope for this specification.

7.1.5.1.4 Actions at a UE receiving a hold offer from a UE not conformant to this specification

A number of variations within SIP implementations exist for how the holding of a media stream should be signaled. A UE that receives an offering SDP that does not conform to this specification needs to respond in a manner compatible with the method invoked by offering party.

If the UE receives an offer SDP with an attribute of a=sendonly in the media block, then the UE MUST answer with an updated SDP that includes an attribute of a=recvonly in the media block of the answer SDP equating to the offered held media.

If a UE receives an offer SDP offer with a media attribute of a=sendonly, the UE SHOULD monitor the received RTP packets and if no far-end media payload is provided, the UE SHOULD provide locally generated comfort noise to the subscriber in order to simulate a held call.

If in addition to an attribute of a=sendonly the offer SDP contains connection data set to c=0.0.0.0, then the UE MUST include connection data of c=0.0.0.0 in the answering SDP.

If the UE receives an SDP offer with no directionality attributes but connection data set to c=0.0.0.0, then the UE MUST answer with an updated SDP to the offering party and include connection data of c=0.0.0.0 in the answering SDP.

If the UE receives an offer SDP with no directionality attributes but connection data set to c=0.0.0.0, then the UE SHOULD provide comfort noise locally to the held subscriber.

7.1.5.1.5 Actions to ensure media keep-alive

When a media stream has been placed on hold via the mechanism defined in this section, no RTP packets are sent between the two UEs as the directionality attribute negotiated is 'inactive'. However, as the media path between the UEs may traverse components such as NATs and/or firewalls, connectivity of the media path needs to be maintained. Media connectivity, however, is maintained through UE conformance to the requirements in [PKT 24.229].

7.1.5.1.6 Actions on retrieval of media

When the controlling UE wishes to retrieve or resume bi-directional media with the held UE, it MUST send an updated offer SDP with an attribute of a=sendrecv in the media block being resumed. A held UE not in a dual-hold state that receives an offer SDP with an attribute of a=sendrecv MUST respond with an updated answer SDP with an attribute of a=sendrecv in the media block being resumed. When a UE that is in a dual-hold state receives a new offer SDP to retrieve the media, then the UE MUST respond with an updated answer SDP with an attribute of a=inactive in the media block of the answer SDP equating to the offered retrieved media.

7.1.5.1.7 RTCP Requirements While On Hold

As noted in [RFC 3264], media directionality has no impact on any negotiated RTCP stream. Thus, RTCP MUST continue to be transmitted to and/or from a UE that is on-hold if RTCP has been negotiated for this session.

7.1.5.2 Feature Data

There is no data associated with this feature.

7.1.5.3 Feature Interactions

See the individual service description sections of this specification for details on the use of held media and any interactions that are present.

7.1.5.4 Call Flows

The call flows in this section illustrate call hold scenarios. The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence. All of the call flows make use of re-INVITEs to provide the updated SDP for the hold and retrieval actions. This is purely illustrative; an UPDATE method may also be used.

Figure 7 illustrates the holding of an existing bi-directional unicast media stream.

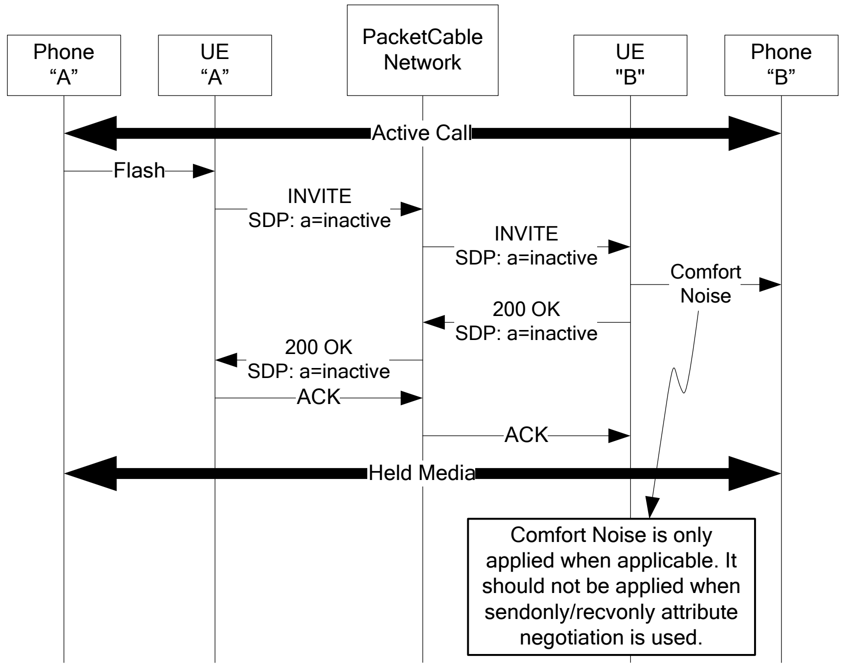

Figure 7 - Sample Held Media Message Flow

Figure 8 illustrates the use of a fallback to [RFC 2543] defined hold when the initial hold attempt fails because the 'held UE' does not understand the a=inactive directionality attribute.

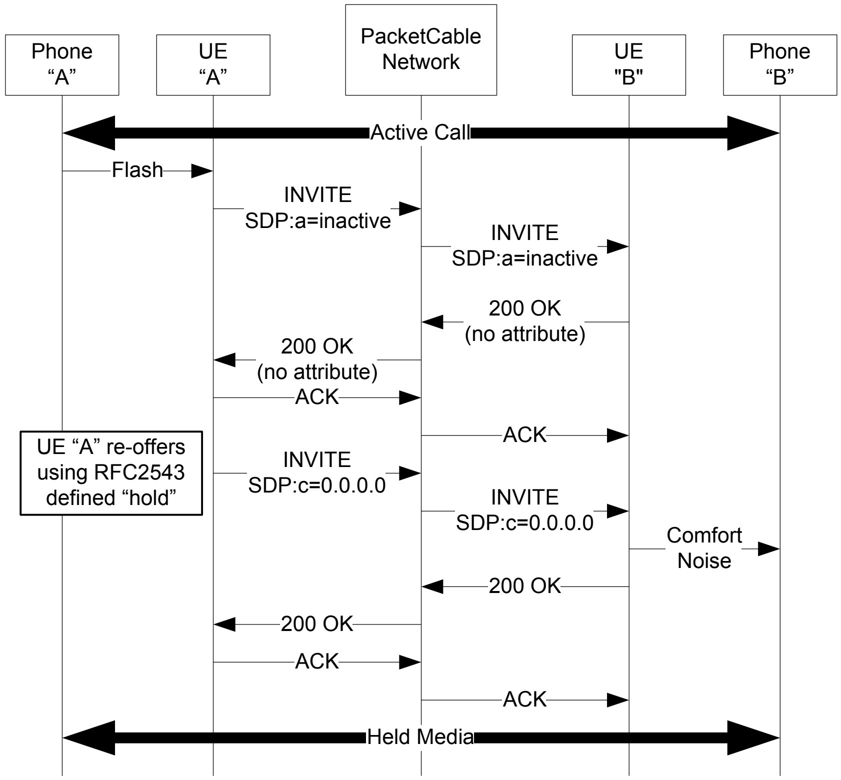

Figure 8 - Sample Held Media Message Flow; Fallback to RFC2543 Defined Hold

Figure 9 illustrates the retrieval of an existing held unicast media stream.

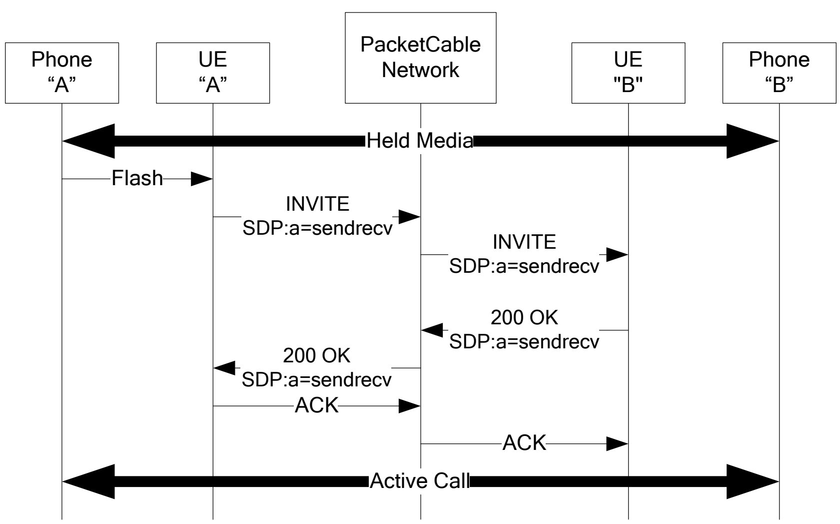

Figure 9 - Sample Held Media Message Flow; Retrieval of Held Media

Figure 10 illustrates a message flow for the transition to a dual-hold state.

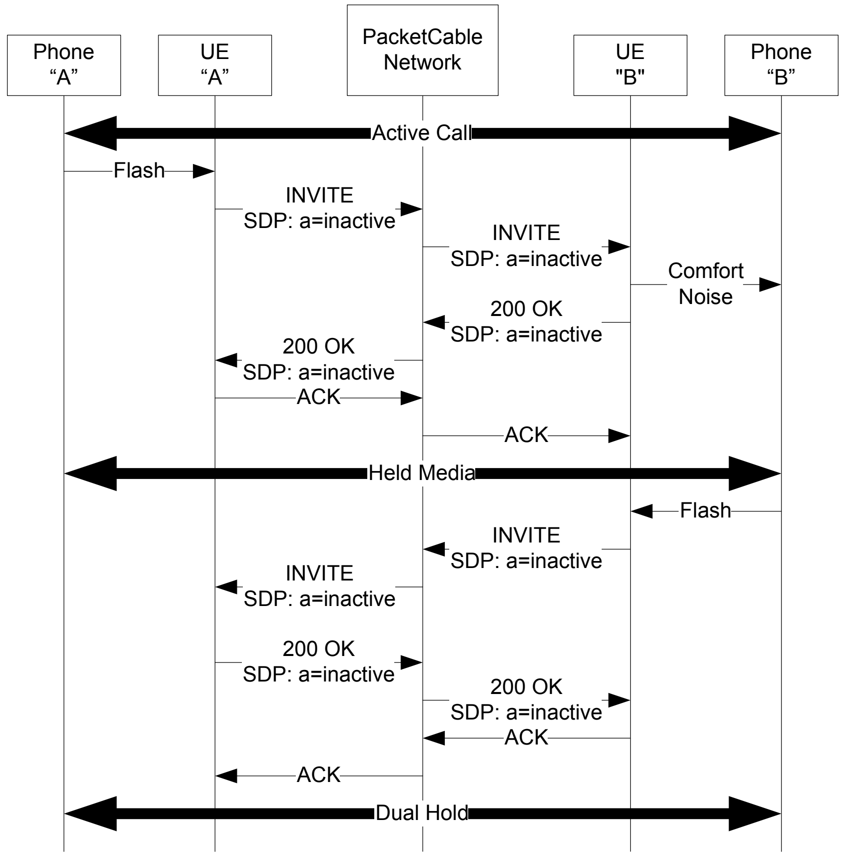

Figure 10 - Held Media; Transition to Dual-Hold

Figure 11 illustrates a message flow when one party in a dual-hold state retrieves the media.

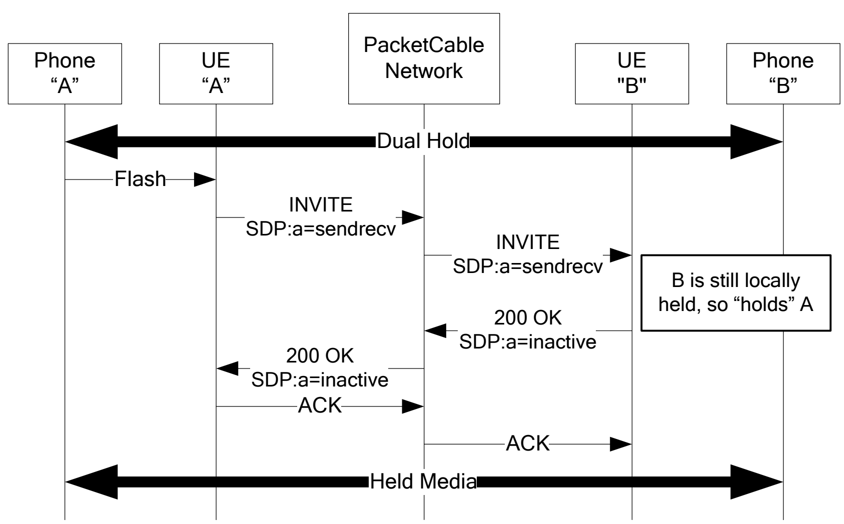

Figure 11 - Held Media; Retrieval by One Party When in Dual-Hold

7.1.6 Announcements

Users are alerted to some network conditions or states by network-initiated announcements. Announcements could be informational tones or messages played to end users in response to events that occur in the network. These announcements could be stored and played locally by the UE to the end user, or they could be received by the UE from a network stream between the UE and a media server.

For the UE, an announcement is defined as the application of an analog signal to the two wire telephony port (RJ-11 analog telephone interface) to provide the end user or device information with respect to a call request or status change. These announcements could be analog audio tones or analog voice or music announcements. Announcements and tones could be provisioned on the UE or on a network announcement source. When a UE needs to present an announcement from a network announcement source, the audio format is negotiated between the UE and the source of the announcement content.

Acoustic or visual announcements by the UE are outside of the scope of this document.

7.1.6.1 Feature Description

North American telephony standards define minimum analog line announcements as a set of tones that are required for specific services or connection status (reference [GR 506], [GR 674], and [GR 675].

In addition to these announcement tones, the UE MAY also be capable of requesting audio announcements to be applied to the analog line from a network announcement application server. The Telcordia [BR 780] Tones and Announcements specification defines text for some possible announcements.

The UE MAY provide audio announcements from an internal, provisioned, audio application server to the analog line.

In addition to the analog line signals, the UE could present acoustic or visual displays equivalent to the tone and voice announcements. Acoustic or visual announcements by the UEs are outside of the scope of this document.

Therefore, there are two types of announcements in PacketCable: tone and audio announcements.

The UE MUST support the analog line tones listed in the PacketCable Network-Based Call Signaling Protocol Specification [NCS].

Audio announcements could be presented by network announcement sources or by audio provisioned locally to the UE. Audio announcements optionally could be provided with a language preference provisioned by subscriber. The use of a local or network announcement source is determined by the UE at the time of announcement.

Local announcements could be used to notify the user that the network is unavailable or is not capable of supporting analog line connections. This specification does not require support for local (UE-resident) audio announcements. Local announcements MUST be provisionable if they are supported.

Network Application Servers provide announcements that are used to notify the user of network level services or requirements. The use of a network server minimizes access device memory requirements and allows for multilanguage support based on the end-user service profile. In general, the procedures described in [PKT 23.218] apply to network announcements, as further described in this section.

7.1.6.2 Activation and Deactivation

The routing of the session requests to an announcement application server is achieved using one of the following methods:

- The application server re-INVITEs the UE to connect the RTP session to the application server.
- The application server redirects the UE to the announcement server using a 3xx Response Code (Reference 7.1.11, Response Codes) during session signaling.
- The application server specifies a network hosted audio treatment using the alert-info header in a response message, allowing the UE to initiate a session. The alert-info header could contain a fully-specified URL.
- The application server sends a 183 session progress response and provides an early media session description that directs the UE to the required announcement.
- If the application server does not provide the announcement, the UE MAY initiate any announcement treatment per the UE provisioning or capabilities, including application of tones.

In response to the INVITE, the AS or Media Server SHOULD query the HSS (Home Subscriber Server) to determine if there are any Subscriber specific requirements (such as language selection) and then provide the UE with the appropriate announcement.

Network-initiated announcement addresses supplied in alert-info headers and contact headers MUST comply with the prompt_url syntax defined by [RFC 4240], SIP Media Services. For example:

```
sip:annc@ms2.example.net;play=http://audio.example.net/allcircuitsbusy.g711
sip:annc@ms2.example.net;play=file://server.example.net//geminii/yourHoroscope.wav
```

The announcement address provided by the Application Server could be a URL. A UE supporting network-based announcements MUST support the file://, sip: and HTTP: URI types. Also, a UE MAY support the FTP URI. The URL provided in the prompt_url uses one of the following URL scheme:

file:/// Indicates that the UE SHOULD look for the announcement media in local storage and play locally

sip: Indicates that the UE SHOULD pass the prompt_url to the configured Media Server in an INVITE request to establish an RTP session for the announcement.
Format is specified in [RFC 3261]

ftp: Indicates that the UE SHOULD download the announcement media from remote storage to play out locally. The UE MAY look for cached announcement media in local storage.
Format is specified in [RFC 2396]

http: Indicates that the UE SHOULD download the announcement media from remote storage to play out locally. The UE MAY look for cached announcement media in local storage.
Format is specified in [RFC 2396]

If the Application Server provides an announcement address using any other URL scheme, the UE MAY forward the announcement address to the configured Media Server in an INVITE request to establish an RTP session for the announcement. Otherwise, the UE applies the default tone or announcement appropriate to the request or response code.

The UE MAY be provisioned to map SIP response codes to announcements. However, to minimize the complexity of the UE, the UE MAY use the SIP Response Code to format the prompt_url in the INVITE message that it sends to the network announcement source. The format of the prompt_url is:

```
sip:annc@<operator_announcement_domain>
play=file://<announcement_network_path>/<response_code>.<announcement_mime_type>
```

The UE needs an operator announcement domain, network path and mime type to allow construction of the prompt_url using this syntax.

7.1.6.3 Tone and Announcement Media

PacketCable supports the following basic categories of announcements, as described in [BR 780]. All network announcements, whether provisioned locally on the UE or referenced to resources in the network, are identified by an announcement identifier. Table 5 describes announcement categories and conditions and the associated Announcement Identifiers.

Table 5 - Announcement Identifiers

Category/Condition	Announcement	Announcement Identifier
Reorder/All Circuits Busy	"Your call cannot be completed at this time. Please hang up and try again later."	Annc-Reorder
Reorder/Protocol Errors	"Your call cannot be completed at this time. Please hang up and try again later."	Annc-Reorder
Vacant Code/No Route	"Your call cannot be completed as dialed. Please check the number and try again."	Annc-Vacant
Intercept/Number Change	Implementation dependent	Annc-NumberChg
Intercept/Disconnected Number	"The number that you have reached has been disconnected. Please check the number and try again."	Annc-Disconnected
Intercept/Nonworking Number	"The number that you have reached is not in service. Please check the number and try again."	Annc-NonWorking
Intercept/Temporary Suspension	Implementation dependent	Annc-TempSusp
Intercept/Incoming Call Restriction	"The number you have reached can not accept your call at this time. Please hang up and try again later."	Annc-Decline
No Circuit	"Your call cannot be completed at this time. Please hang up and try again later."	Annc-NoCircuit
Ineffective/Dialing Irregularity	"Your call cannot be completed as dialed. Please check the number and try again."	Annc-DialError
Ineffective/Screened Line Access Denial	Implementation dependent	Annc-AccessDenied

As described previously, the UE MAY be provisioned to identify, locate, and play announcements based on the presence of Alert-info headers or Contact headers provided by other network elements in response messages, or simply by the response code if no announcement is requested. Table 6 relates final response codes (4xx, 5xx, 6xx) to announcements identifiers.

Table 6 also identifies announcements that only can be played if additional information is provided by a network server by including an Alert-info header or Contact header in the response message. For example, the response 487 Request Terminated, by default, invokes the reorder announcement, if provisioned. A network server can initiate a more detailed announcement, when available, by specifying the appropriate Announcement Identifier in a contact header.

Table 6 - Announcement Map

Response Code	Announcement Identifier
404 Not Found	Annc-Vacant
406 Not Acceptable	Annc-Reorder
408 Request Timeout	Annc-Reorder
480 Temporarily Unavailable	Annc-Reorder
484 Address Incomplete	Annc-DialError
487 Request Terminated, with no Contact Header	Annc-Reorder

Response Code	Announcement Identifier
487 Request Terminated, with Contact Header specifying Annc-Disconnected	Annc-Disconnected
487 Request Terminated, with Contact Header specifying Annc-NonWorking	Annc-NonWorking
500 Server Internal Error	Annc-Vacant
503 Service Unavailable	Annc-Vacant
504 Server Time-out	Annc-Vacant
600 Busy Everywhere	Annc-Decline
603 Decline	Annc-Decline

The UE MUST use the received response code to determine whether or not an announcement has been provisioned, by examining the Announcement Map specified in Table 6 for a matching entry. When there is a matching entry, the UE MUST retrieve the Announcement Identifier from the entry and use this to retrieve the Announcement URI from the Announcement Media Map specified in Table 7. The Announcement URI MAY specify a locally stored announcement, tone, or bell cadence by using the file syntax, in which case the UE MUST look up the alert media in its Local Media Table (see Table 8).

Figure 12 illustrates this process.

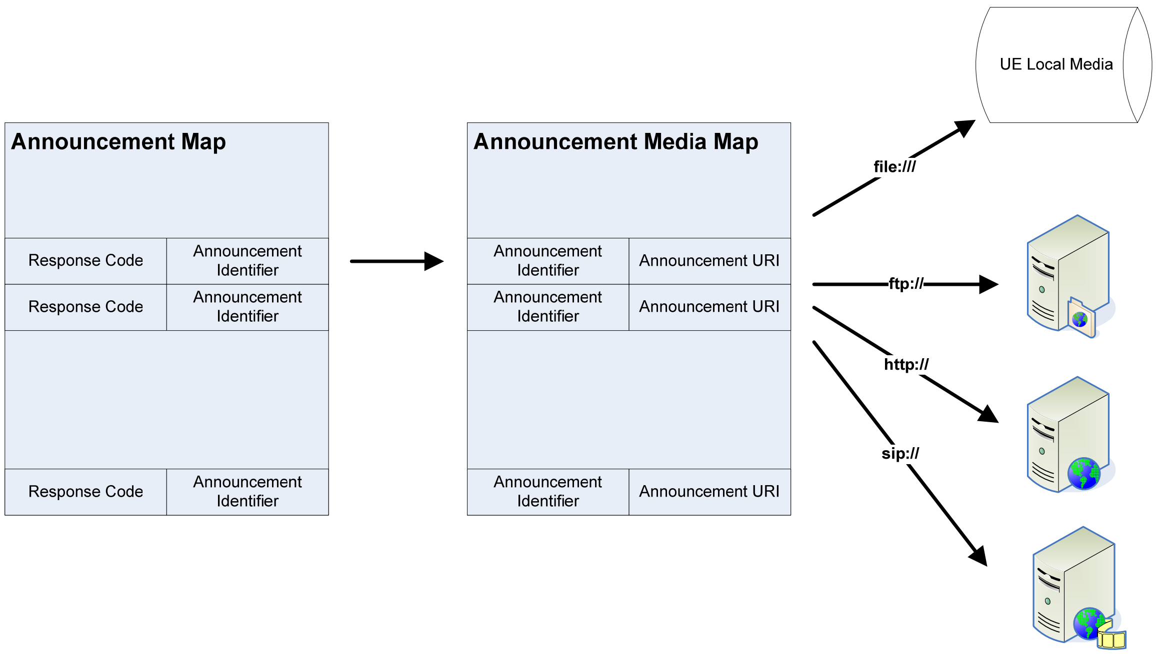

Figure 12 - Announcement Media Mapping

When there is no entry in the Announcement Map for a response code, the UE MAY construct a URI following the procedure described in Section 7.1.6.2.

7.1.6.4 Management of Announcement Media Cache

As described in Section 7.1.6.2, the UE MAY cache announcement media fetched using FTP and HTTP in local storage. Caching announcement media locally provides obvious performance advantages, but when announcement media is locally cached, there is no assurance that the cached version is the most current. Even though the announcements typical of a telephony system probably do not change very often, the UE should not keep announcement media cached indefinitely.

HTTP deals with the aging of cached resources with headers that allow clients and servers to specify expirations. The UE can make use of these protocol elements. FTP does not have similar protocol elements. The UE also can control cache contents independently of the protocols.

UE Cache Control

If the UE caches network announcement media, then the UE MUST support a configurable parameter for the maximum-age of any cached announcement media. This one parameter applies to all cached media fetched by FTP and HTTP. When the UE needs to access cached announcement media that has aged beyond the configured maximum-age for cached resources, the UE MUST fetch it from the network media server. The UE MAY discard cached announcement media that has aged beyond the configured maximum-age for cached resources.

HTTP Cache-Control

HTTP provides the Expires header and the Cache-Control header to help control cache contents. Media Servers SHOULD use the HTTP Expires header to indicate how clients should treat the announcement media. Media servers MAY use the cache-response-directive of the Cache-Control header for this purpose. If the UE caches network announcement media, then the UE MUST NOT use the Cache-Control header in its HTTP requests. The UE manages its cache contents for announcement media retrieved by HTTP for specific values of the Expires header and Cache-Control header as follows.

Expires Header and Max-Age Directive

When the media server specifies the Expires header or max-age directive in a Cache-Control header, and the UE is caches network announcement media, then the UE MUST compare the expiration time/max-age received for the announcement media with the configured maximum-age for its media cache. If the resource's expiration/max-age is less than the cache's maximum-age, the UE MUST invalidate the resource after it has aged as defined by the server. If the media server provides both an Expires header and a max-age directive, the UE MUST apply the more restrictive value.

No-Store Directive

When the media server specifies the no-store directive in a Cache-Control header, and the UE caches network announcement media, then the UE MUST NOT store the announcement media in its cache. After playout, the media MUST be discarded.

7.1.6.5 Feature Execution

This section describes various scenarios for network initiated announcements. Each scenario starts with the UE originating a call using basic SIP signaling messages. The scenarios describe the ways network-initiated announcements are executed, given that the network elements could initiate announcements in several ways.

Announcement Address Not Provided by Application Server

The Application Server may not support any audio announcement capabilities for some situations that can be handled by other means. The UE responds by playing the default tone, a local announcement or a network announcement, depending on its provisioning:

> *Default Tone*. In the absence of any network announcement provisioning, the UE MUST play the default tone defined for the response code received.
>
> *Local Announcement*. If the UE was provisioned with announcement audio media for the received SIP response code, the UE MUST play this audio announcement file.

Network Announcement. If the UE is provisioned with network announcement path and type, the UE MUST attempt to play the provisioned announcement by response code by constructing a prompt_url for the announcement address and sending an INVITE to the provisioned network announcement source to establish a media session.

The network announcement source plays the required announcement. When the playing of the announcement is complete, the network announcement source ends the media session by sending a BYE to the UE.

If the announcement source rejects the UE INVITE message (for example, the announcement source can not map the prompt-url constructed by the UE to a valid announcement media), the UE MUST apply the default tone or announcement appropriate to the request or response code.

Announcement Address Provided by Application Server in Alert-info Header

The Application Server may provide an announcement address in an alert-info header returned with a final response message.

- The application server provides the UE with the announcement address as a prompt_url in the alert-info header supplied with a response message.
- The UE originates an INVITE to the provisioned network announcement source to establish a media session.
- The UE performs normal negotiation with the network announcement source to establish the media session.
- The network announcement source plays the required announcement.
- When the playing of the announcement is complete, the network announcement source ends the media session by sending a BYE to the UE.

Announcement Address Provided by Application Server in Contact Header

The Application Server may provide an announcement address in a Contact header returned with a final response message.

- The application server provides the UE with the announcement address as a prompt_url in the Contact header supplied with a response message.
- The UE originates an INVITE to the provisioned network announcement source to establish a media session.
- The UE performs normal negotiation with the network announcement source to establish the media session.
- The network announcement source plays the required announcement.
- When the playing of the announcement is complete, the network announcement source ends the media session by sending a BYE to the UE.

Announcement Address Provided by Application Server in Redirect Response

The Application Server may provide an announcement address in a Contact header returned with a redirect response message. The UE proceeds as described in the preceding section.

Announcement Provided by Application Server in Early Media Response

The Application Server may provide the audio announcement itself, and reflect this by returning a 183 Call Progress response message with SDP to the UE. The UE responds by negotiating the media session with the server and playing the audio supplied by the server. In this case, no final response has been generated. When the server network announcement source has completed playing audio, the server source sends a final response message of 487 Request Terminated.

7.1.6.6 Feature Data

Table 7 summarizes the feature data elements defined to support implementation of the Announcement feature.

Table 7 - Announcement Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Announcement Resource	URI for media server	Non-volatile on AS and XDS, Volatile on UE	Per network	XDS	XDS	AS, UE	Mandatory
Announcement Domain	Domain name	Non-volatile on AS and XDS, Volatile on UE	Per network	XDS	XDS	AS, UE	Mandatory
Announcement Path	Network file path	Non-volatile on AS and XDS, Volatile on UE	Per network	XDS	XDS	AS, UE	Mandatory
Announcement mime type	Mime type (text)	Non-volatile on AS and XDS, Volatile on UE	Per network	XDS	XDS	AS, UE	Mandatory
Announcement map	Variable length array of Announcement map entries	Non-volatile on XDS, Volatile on UE	Per network	XDS	XDS	UE	Mandatory
Announcement map entry.Response code	Integer	Non-volatile on XDS, Volatile on UE	Per network	XDS	XDS	UE	Mandatory
Announcement map entry.Annoucem ent URI	String identifying uri for response code	Non-volatile on XDS, Volatile on UE	Per network	XDS	XDS	UE	Mandatory
Announcement media map	Variable length array of Announcement Media Map entries	Non-volatile on XDS, Volatile on UE	Per network	XDS	XDS	UE	Mandatory
Announcement media map entry.Announce ment Identifier	String identifying the announcement	Non-volatile on XDS, Volatile on UE	Per network	XDS	XDS	UE	Mandatory
Announcement media map entry.Announce ment URI	String identifying uri for Announcement Identifier	Non-volatile on XDS, Volatile on UE	Per network	XDS	XDS	UE	Mandatory
Announcement media cache maximum-age	Integer (Seconds)	Non-volatile on XDS, Volatile on UE	Per Network	XDS	XDS	UE	Optional

Table 8 summarizes the feature data elements defined to support provisioning of local media on the UE.

Table 8 - Local Media

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Local Media Table	Variable length array of Media entries	Non-volatile on XDS, Volatile on UE	Per network	XDS	XDS	UE	Mandatory
Media.URI	File:/// URI identifying locally stored media	Non-volatile on XDS, Volatile on UE	Per network	XDS	XDS	UE	Mandatory

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Media.Type	Media Type (from set of bell cadence, tone generator specification, or audio file)	Non-volatile on XDS, Volatile on UE	Per network	XDS	XDS	UE	Mandatory
Media.Data	The actual media data, format depends on Media.Type	Non-volatile on XDS, Volatile on UE	Per network	XDS	XDS	UE	Mandatory

Local media types supported are audio files, ring cadences or tones. Audio files are identified by the file type (for example, mp3, wav) are implementation dependent. Ring cadences and tones are defined by [ID IPCDN NCSMIB].

- Ring cadences are defined by instances of the PktcRingCadence object in the pktcSigDevRingCadenceTable.
- Single frequency tones are defined by instances of pktcSigDevToneEntry in the pktcSigDevToneTable.
- Multiple frequency tones are defined by instances PktcSigDevMultiFreqToneEntry in the PktcSigDevMultiFreqToneTable.

7.1.6.7 Feature Interactions

To the extent that features need announcements to be played, each feature invocation initiates the announcement processing.

Announcement processing interaction with the UE is not recorded in the last outgoing call memory or last incoming call memory (used for recording the most recently placed call and the most recently received call, respectively), in order to protect the integrity of the Auto Recall, COT, and Auto Callback features.

7.1.6.8 Support for Language Preference (Optional)

PacketCable may optionally support user language preference. This support is provided through several mechanisms:

- The UE MAY provide hints of language preference to network elements using the locale feature tag in the Contact header in basic call signaling that establishes dialogs;
- Servers that support network-initiated announcements MAY use hints received from UE or language preferences provisioned in the subscriber's profile to play an announcement in the subscriber's preferred language;
- Servers that provide announcement addresses to UE MAY provide multiple language choices to allow the UE to select the most appropriate presentation.

These capabilities are described in the following subsections.

7.1.6.8.1 Syntax of Announcement Addresses for Language Preferences

UEs indicate language capabilities to application servers using one or more Contact headers in all dialog-establishing SIP requests and responses. Application servers indicate language preference by either directing the UE to the appropriate announcement address as described in Section 7.1.6.2, or by providing an [RFC 4240] compliant prompt_url with *locale* parameter in a Contact header.

When the UE includes language preferences in Contact headers, it MAY include a Contact header with a locale parameter listing the preferred languages, using the format of local tags as defined by [RFC 3066]. For example:

```
Contact: <sip:3035551212@ms2.example.net;user=phone; \
locale="en-us,es-us,fr-ca">
```

When the application server adds a *locale* parameter to the prompt_url, the application server also MUST use locale tags as defined by [RFC 3066]. For example:

```
Alert-info: <sip:annc@ms2.example.net; \
       play=http://audio.example.net/<response_code>.g711;locale=en-us>
```

When the UE receives an error response without any indication for network announcement, with or without language preference, and if the UE is configured appropriately, the UE MAY construct a prompt_url for a network-based announcement using the received response code. Using response codes to the Announcement Server to "map" to the desired announcements MAY be represented as any one of the following:

```
Contact: <sip:annc@ms2.example.net; \
       play=http://audio.example.net/<response_code>.g711
Contact: <sip:annc@ms2.example.net; \
       play=http://audio.example.net/<response_code>.g711;locale=en-us>
Contact: <sip:annc@ms2.example.net; \
       play=http://audio.example.net/<response_code>.g711;locale=es-us>
Contact: <sip:annc@ms2.example.net; \
       play=http://audio.example.net/<response_code>.g711;locale=fr-ca>
```

If an application server redirects the UE to an announcement server using a 3xx redirect, then the application server MAY provide multiple contacts to the UE, allowing the UE to pick which of the contacts to use based upon its language preferences. For example, an application server could provide all of the above contacts to the UE.

7.1.6.8.2 Behavior of UE for Language Preferences

The UE MAY be provisioned to indicate one or more optional subscriber language preferences. When provisioned for more than one language, the default language MUST be configured.

The UE MAY use the provisioned language preferences to play the appropriate announcement in response to these events:

Receipt of a basic SIP response message. When the UE receives a SIP response message that does not include any indication of network announcement to be played, the UE constructs its INVITE for the announcement as described in 7.1.6.5. To indicate a language preference, it SHOULD include a *locale* parameter as described in [RFC 4240].

Receipt of SIP re-INVITE, redirect or response message with announcement address. When the UE receives a SIP re-INVITE, redirect or response message with one announcement address in the form of a prompt_url, per [RFC 4240], the UE MUST process these messages as described in Section 7.1.6.2. If the UE is provisioned with a language preference, and if the prompt_url does not contain a *locale* parameter, then the UE MAY update the prompt_url to include a *locale* parameter for its configured (default) language preference.

Receipt of SIP re-INVITE, redirect or response message with multiple announcement addresses. When the UE receives a SIP re-INVITE, redirect or response message with multiple announcement addresses, the UE processes the message as follows.

When the UE receives a SIP re-INVITE, redirect or response message with multiple announcement addresses and the UE is not provisioned with language preference, then the UE MUST process the message as described in Section 7.1.6.2.

When the UE receives a SIP re-INVITE, redirect or response message with multiple announcement addresses and is provisioned with language preference, then the UE MAY scan the offered announcement addresses, and reorder the announcement addresses, placing its provisioned language preference first. When the UE does not support this operation, the announcement address placed first by the AS is interpreted as the default language preference.

When the UE is processing a SIP response message, as with any announcement, after the UE has applied optional language preference processing, the UE MAY play a locally stored announcement if this is otherwise consistent with the announcement address.

7.1.6.8.3 Feature Data for UE Announcement Language Preference

Table 9 summarizes the feature data items that are defined to support the announcement language preference feature.

Table 9 - UE Announcement Language Preference Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Language	Character String	Non-volatile in XDS, volatile in UE	Per public user ID	XDS	XDS	UE	Mandatory

7.1.6.8.4 Behavior of Application Server for Language Preference

All language preference support by network elements such as an Application Servers is optional. An Application Server MAY indicate language preference in several ways:

- Language-specific Announcement Addresses:

  Application Servers MAY provide language-specific announcements by simply redirecting the UE to the appropriate announcement address, or providing the appropriate announcement address in the response message. The method of determining the correct announcement address is application-dependent, and MAY use the subscriber's profile or UE hints.

- Subscriber profile preference:

  The Application Server MAY include Contact headers containing prompt_urls with locale parameters in SIP response messages.

- Contact Headers:

  Application Servers MAY include Contact headers in redirect and response messages to indicate language preference. Each Contact header specifies an announcement address in [RFC 4240] prompt_url format, with locale parameter, as previously described, in preference order. As with Accept-Language headers, the value of the Contact header locale parameters MAY be derived from provisioning, from the subscriber profile, or from UE hints. A UE indicates its language preference by including a locale parameter in its contact header.

7.1.6.8.5 Behavior of Network Media Server for Language Preference

A Network Media Server MAY store announcement media in multiple languages. When multiple languages are supported, the Network Media Server selects the announcement media to play in one of the following ways. Network Media Servers play announcements specified completely by the [RFC 4240] prompt_url received from an Application Server or UE. Network Media Servers MAY select an announcement to play based on the locale parameter received from the UE or the Application Server.

When the network Media Server completes playing an announcement, the network server's behavior depends on whether it is an early media session. If the media session is not early media, the network Media Server ends the session by sending a BYE to the UE. If the media session is early media, the network Media Server ends the session by sending a 487 Request Terminated message to the UE.

7.1.6.9 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

Figure 13 illustrates a call flow involving an Application Server returning an error final response with an announcement address specified in an alert-info header. In response, the UE sets up a session with a Media Server.

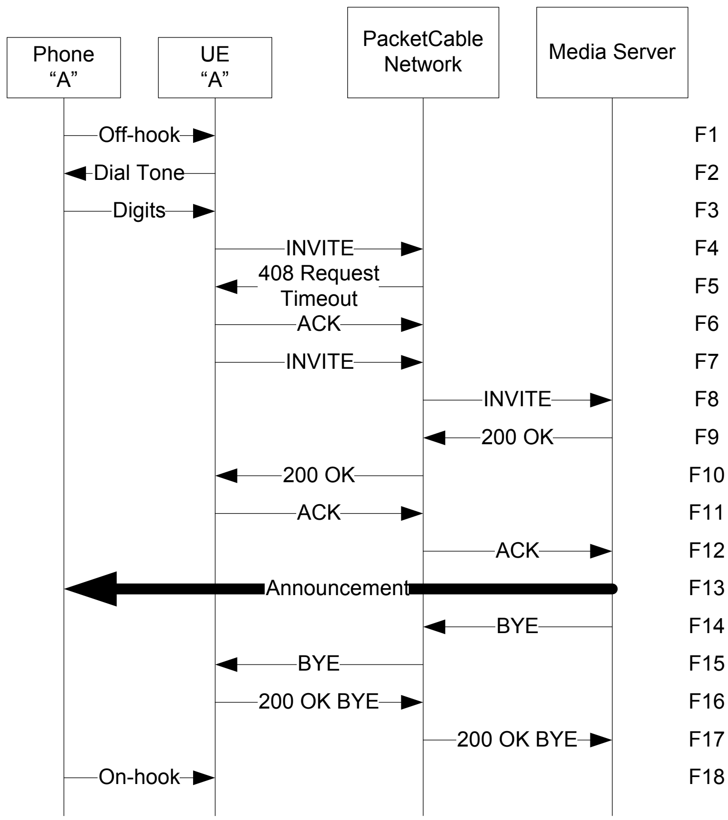

Figure 13 - Media Serving; UE INVITE for Network Announcement

```
F5 408 REQUEST TIMEOUT   UE <- SIP NETWORK

SIP/2.0 408 REQUEST TIMEOUT

Contact: <sip:annc@ms2.example.net; \
       play=file://operator.anncserver.net//audio/408.wav>

F7 INVITE   UE -> SIP NETWORK

INVITE sip:annc@operator.anncserver.net;play=file:///408.wav SIP/2.0
```

Figure 14 illustrates an Application Server redirecting the UE to the Media Server. The underlying reason for the redirect is not reflected in the response code. Instead, Application Server provides the reference to the appropriate announcement in a Contact header.

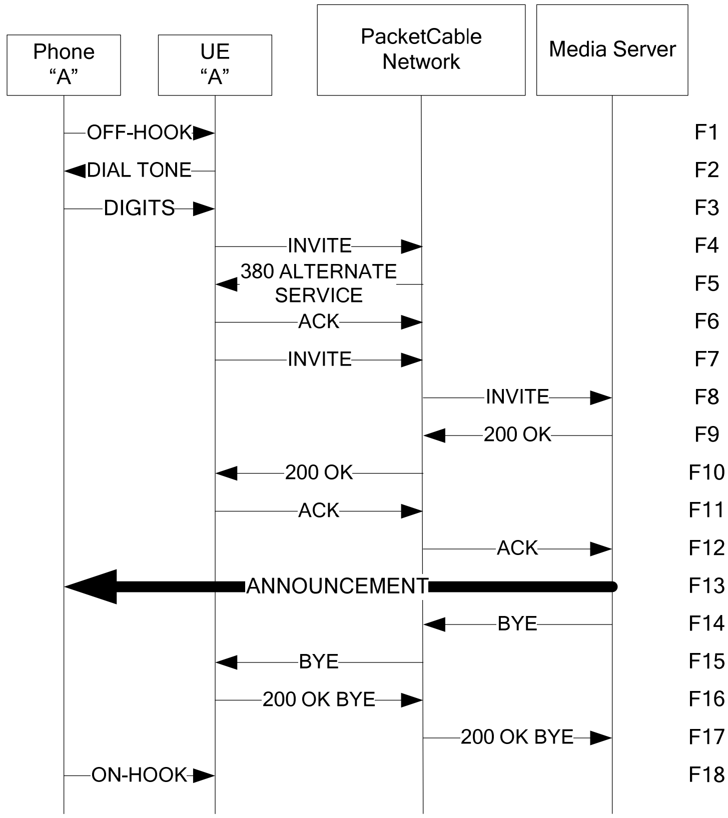

Figure 14 - Media Serving; Server Redirecting UE

```
F5 380 ALTERNATE SERVICE   UE <- SIP NETWORK

SIP/2.0 380 ALTERNATE SERVICE
Contact: <sip:annc@ms2.example.net; \
       play=file://operator.anncserver.net//audio/408.wav>

F7 INVITE   UE -> SIP NETWORK

INVITE sip:annc@operator.anncserver.net; \
play=file://operator.anncserver.net//audio/408.wav SIP/2.0
```

Figure 15 illustrates an Application Server simply connecting the UE to an Announcement Server using early media SDP returned with a 183 Session Progress response.

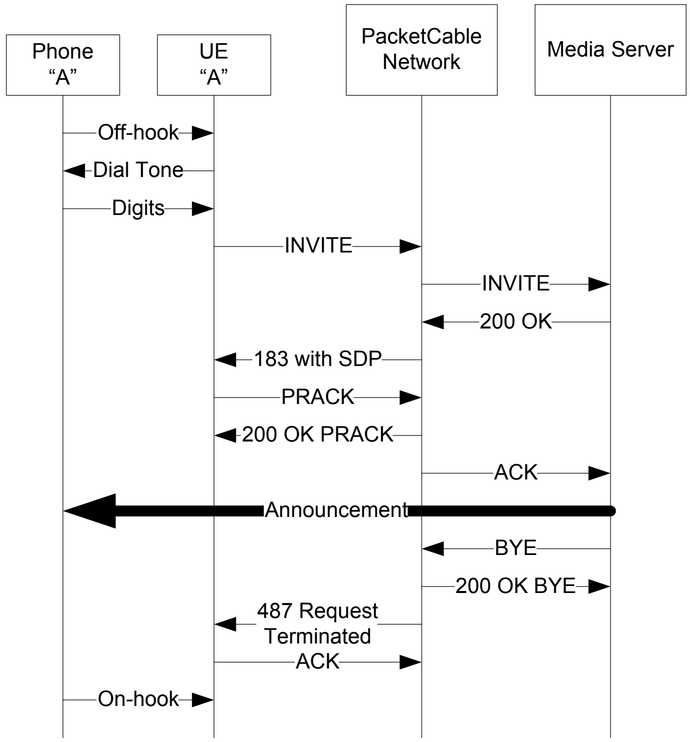

Figure 15 - Media Serving; Server Providing Announcement with Early Media

Figure 16 illustrates an Interactive Voice Response application in which the IVR server accepts spoken word from the user and plays additional prompts by use of re-INVITE. This type of operation might also be invoked by an Application Server that does not have announcement capability itself, and during processing of user interactions needs to play an announcement.

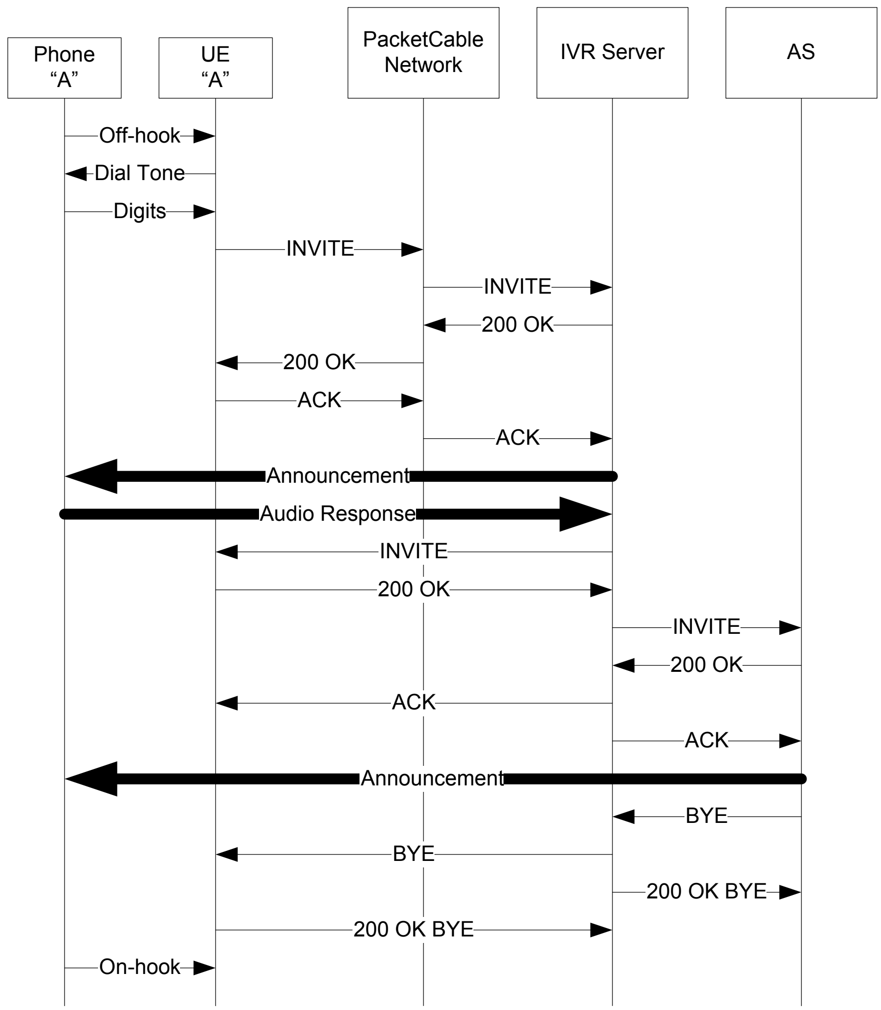

Figure 16 - Media Serving; Network Announcement by Re-INVITE from SIP Network

Figure 17 illustrates how a UE selects an announcement address from those offered by an Application Server. In this example, assume that the UE has been provisioned with American English as its preferred announcement language.

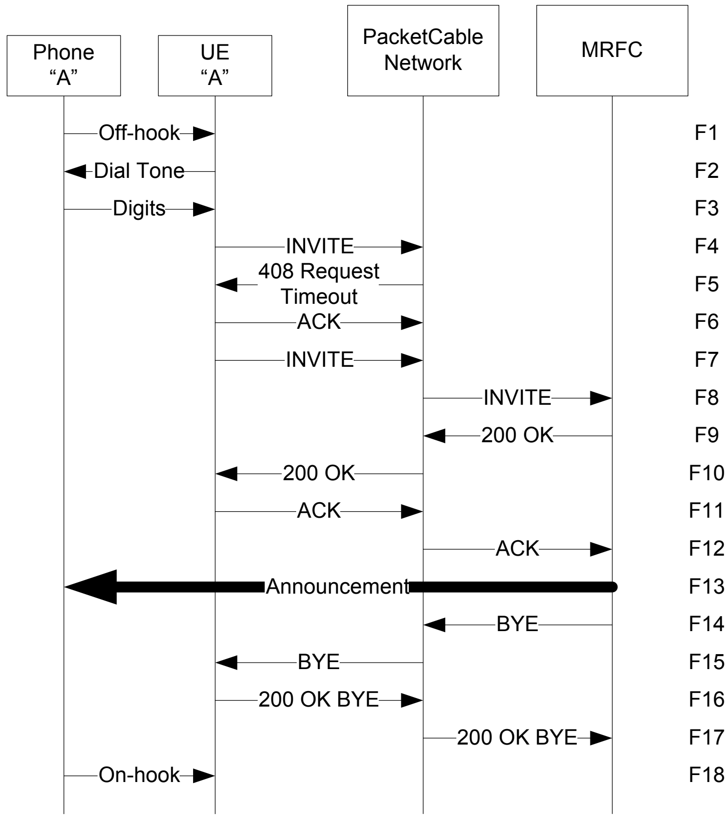

Figure 17 - Media Serving; UE INVITE for Network Announcement with Language Preference

```
F5 408 REQUEST TIMEOUT   UE <- SIP NETWORK

SIP/2.0 408 REQUEST TIMEOUT
…
Contact: <sip:annc@ms2.example.net; \
  play=file://operator.anncsvr.net//audio/408.wav;locale="en-us,es-us,fr-ca">

F7 INVITE   UE -> SIP NETWORK

INVITE sip:annc@ms2.example.net; \
  play=file://operator.anncsvr.net//audio/408.wav;locale=en-us SIP/2.0
```

7.1.7 In-Service and Out-of-Service States

7.1.7.1 Feature Description

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 for most of us have led us to believe that no dial-tone always means that the telephone is out-of-service. SIP 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 SIP UE operation would call for the registered SIP 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, P-CSCF). Thus, there is no behavioral shift required for users moving from traditional PSTN to PacketCable services.

7.1.7.2 Feature Execution

A registered UE MUST indicate it is 'in-service' state if it has successfully exchanged a Binding Request/Response with the P-CSCF STUN server, and, if STUN keepalive messages are being exchanged between the UE and the STUN server located within the P-CSCF as defined in [ID SIP-OUTBOUND]. This indication is in the form of the dialtone, which is provided when the user goes off-hook. Otherwise, the UE MUST assume it is in the 'out-of-service' state and not provide dial tone after a user goes off-hook.

The mechanism chosen to indicate UE connectivity to the SIP Network is derived from [ID SIP-OUTBOUND] and it employs STUN "keep alive" messages between the UE and the STUN server located within the P-CSCF. Figure 18 shows the UE connected to the P-CSCF through a NAT/Firewall. The STUN Server shown as an instance in each P-CSCF is part of the PacketCable architecture to traverse NAT/Firewalls by creating multiple "service flow" bindings between the UE and P-CSCF. The PacketCable approach to NAT traversal is described in [NFT TR]. These bindings are created when the UE registers. Since only the STUN server is employed by the in-service/out-of-service feature, the entire ICE/STUN/TURN issues of NAT traversal are not discussed; only the necessary STUN functionality is described. ICE is described in [ID ICE] and TURN is described in [ID-BEHAVE-TURN].

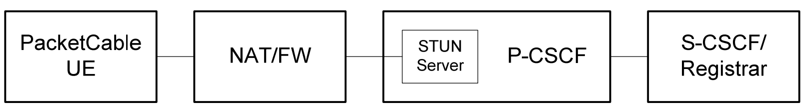

Figure 18 - Functional Components for Determining In-Service/Out-of-Service State

The ICE/STUN/TURN approach to NAT traversal has not yet been adopted by 3GPP for IMS. This requires that changes be made to the 3GPP specifications for PacketCable. These changes are documented in the following PacketCable delta specifications:

- [PKT 23.228] includes requirements for supporting NAT and Firewall traversal.
- [PKT 24.229] provides procedures related to signaling and media traversal of NAT and Firewall devices.

The keep-alive mechanism used to maintain the outbound connection from the UE to P-CSCF is described in [ID SIP-OUTBOUND]. The key concept of this draft is when the UE sends a REGISTER request to the Registrar (via the P-CSCF); the P-CSCF can use the same connection for communication in the other direction (network toward the UE). The draft is called "SIP outbound" because the UE initiates and maintains this connection (referred to as a "flow" in the specification). The UE must also determine when a connection or flow has failed, hence the need for the keep-alive messages.

The STUN protocol is used as a keep-alive mechanism so that one or more flows to the proxy (P-CSCF) and registrar (S-CSCF) remain alive and connected to the SIP Network. It is assumed that the UE is registered with the S-CSCF as described in [PKT 24.229]. As such, this mechanism provides a logical indication for the UE to determine if it is appropriate to provide dial-tone when the user goes off hook.

To provide the in-service/out-of-service indication, the UE MUST:

- Provide the locally-generated dial-tone when the caller initiates a call session by going off-hook or otherwise indicates that a call is being initiated.
- Inhibit providing the dial-tone when no response is received from STUN keep-alive messages consistent with [ID SIP-OUTBOUND] from any established flow.
- Provide locally-generated re-order tone to the caller if network connectivity (indicated by the keep-alive messages) does not exist when a call is initiated.

The complete ICE/STUN/TURN NAT traversal process employed by PacketCable is beyond the scope of this specification. This feature simply uses the keep-alive mechanism of STUN as an indicator of network connectivity. Also not described here are the steps a UE would take to re-register once network connectivity has been lost.

7.1.7.3 Feature Data

The in-service/out-of-service feature indication is by definition UE-based and it is always activated, eliminating the need to provision the function from the network.

7.1.8 UE Status Change

UE status change is the method of indicating to the network that the UE is available to make and receive calls. Typically, UE availability is determined by the registration status of the UE as held by the S-CSCF. The registration update mechanism is determined by the Border controlling elements of the Network. This is typically by accepting REGISTER from the device with the subscriber's authorization information. For more detail on this process, please refer to the PacketCable's 3GPP Security document [PKT 33.203].

7.1.8.1 Feature Execution

Authentication of the UE is required for validation of the subscriber in the Network. This affects both the originating and terminating service invocation. See [PKT 24.229], and [PKT 33.203] for the detailed requirements for authentication. The UE MUST require a valid registration for origination for all features except for configuration.

The specification [PKT 24.229] defines the default registration duration. The Network may decide to change the registration time for a UE. The mechanism for this is either via PACM or the last registration reply from the Network.

- If the Registration Expiration (Table 10) has been configured on the UE through PACM, the UE MUST use it as the expiration duration requested for any non-refresh registration attempt.
- Upon receipt of a registration response indicating that the requested expiration interval is too brief (a 423 response), the UE MUST use the minimum registration expiration interval value provided in the 423 response only for the current registration session. The UE MUST NOT use the duration received in the registration response upon the next non-refresh registration attempt.

7.1.8.2 Feature Data

In addition to registration data required by [PKT 24.229], the feature data shown in Table 10 is defined to support implementation of the UE Status Change feature.

Table 10 - UE Status Change Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Registration Expiration	Integer (in seconds)	Non volatile	Public Identity	XDS	XDS	UE	Mandatory

7.1.9 Early Media

Early media is media exchanged before the call's full dialog is established, as described by [RFC 3960]. The media stream may be one or two-way. The usual example of one-way media is ringback: A calls B, A hears ringtone from B via early media even before the SIP dialog is confirmed (which happens when B answers the call). An example of two-way media is solicitor blocking: A calls B, A is connected to an application server for collection of a greeting. The media stream is bi-directional; one stream to the application server for greeting record, the other from the application server to request the greeting. In both examples the call is not considered established as the SIP dialog initiating transaction is not complete.

In addition to establishing the media stream prior to establishing the session dialog, early media may involve several intermediate established media sessions called early session media. This form of media is necessary for features like Solicitor Blocking, Outbound Call blocking, and anytime media is established with a secondary source prior to connection to the final terminator.

7.1.9.1 Feature Execution, Early Media from Terminating Party

Early media execution involves two parties in a basic call: the originator, who receives early media from the terminator upon receiving an 18X provisional response, and the terminator, who receives early media after sending a positive response to a PRACK from the originator.

A UE MUST support PRACK as defined in [RFC 3262].

For the originator in the call:

- Early media MAY be received by the originator at any time after receiving an 18X provisional response from the terminator. The 18X provisional response for early media MUST include '100rel' in the Require header if the originating party indicated support for PRACK.
- The UE's application of early media based on the reliable response MUST be in accordance with the rules defined in Section 7.1.4.1
- Renegotiation of media by the UE MAY occur any time during an early dialog via UPDATE from the terminator.

For the terminator in the call:

- The terminating UE receiving a compatible SDP in the initial INVITE MUST send an 18X response with the negotiated CODEC.
- When the terminating UE receives a PRACK, it SHOULD begin sending any available media to the originating UE. The terminating UE MAY begin sending any available media to the originating UE before receiving a PRACK.

- Renegotiation of media by the UE MAY occur any time during an early dialog via UPDATE from the originator.

7.1.9.2 Feature Execution, Early Media from Intermediate Parties

Early session media may be used for playing announcements as an intermediate step before terminating to a UE. Examples include Solicitor Blocking, Outbound Call Blocking, etc.

For the originator in the call, in addition to normal Early Media execution:

- UEs supporting early-session MUST support early-session answer of an early-session offer

For the terminator in the call that is providing early media, in addition to normal Early Media execution:

- The call terminator (usually AS) SHOULD support early session media offer answer. The call terminator (usually AS) MUST send an early-session offer when the originator signals it supports early-session via population of the Supported header in the initial INVITE.

7.1.9.3 MGC Considerations

For calls placed from the PSTN to a PacketCable network, the MGC plays the role of the originating UE in early media scenarios. The MGC requirements and procedures (including SIP to ISUP interworking) for support of early media are specified in [CMSS1.5].

7.1.9.4 Feature Data

There is no data associated with this feature.

7.1.9.5 Feature Interactions

Basic Call

- The UE SHOULD render early media as described in Section 7.1.9.1 when early media is received, in lieu of local ringback.

DND PIN Override

- DND PIN override uses two-way early media as described in Section 7.1.9.2 for announcement, and collection of PIN. The DND AS MUST support [RFC 3959].

Solicitor Blocking

- Solicitor Blocking uses two-way early media as described in Section 7.1.9.2 for both the collection of greeting and the playback of greeting to the subscriber. The SB AS MUST support [RFC 3959].

Outbound Call Blocking

- Outbound Call Blocking uses two-way early media as described in Section 7.1.9.2 for collection of overriding PIN. The OCB AS MUST support [RFC 3959].

7.1.9.6 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

Figure 19 illustrates setup of two-way early media from terminating party, with one-way media as an intermediate step.

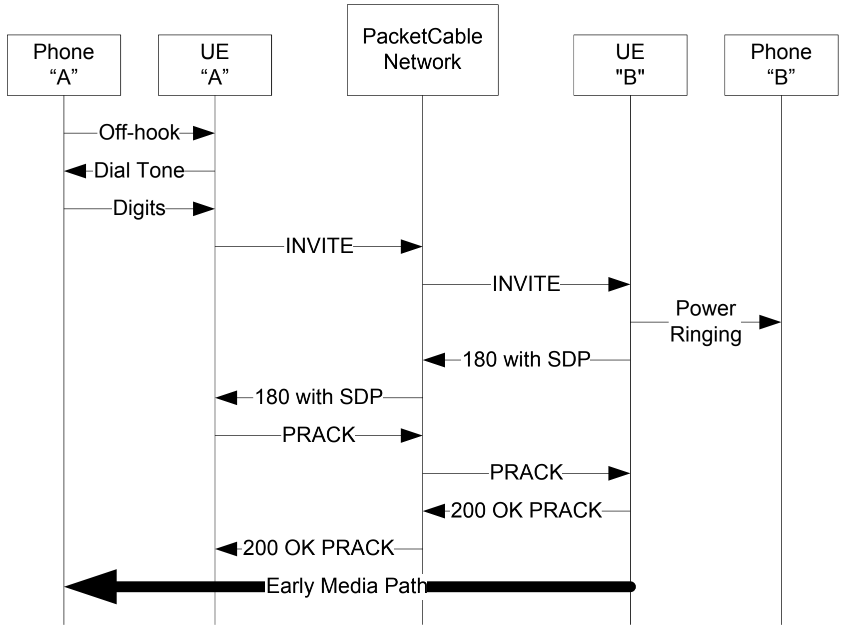

Figure 19 - Early Media from terminating party

Figure 20 illustrates setup of two-way early media from an intermediate party.

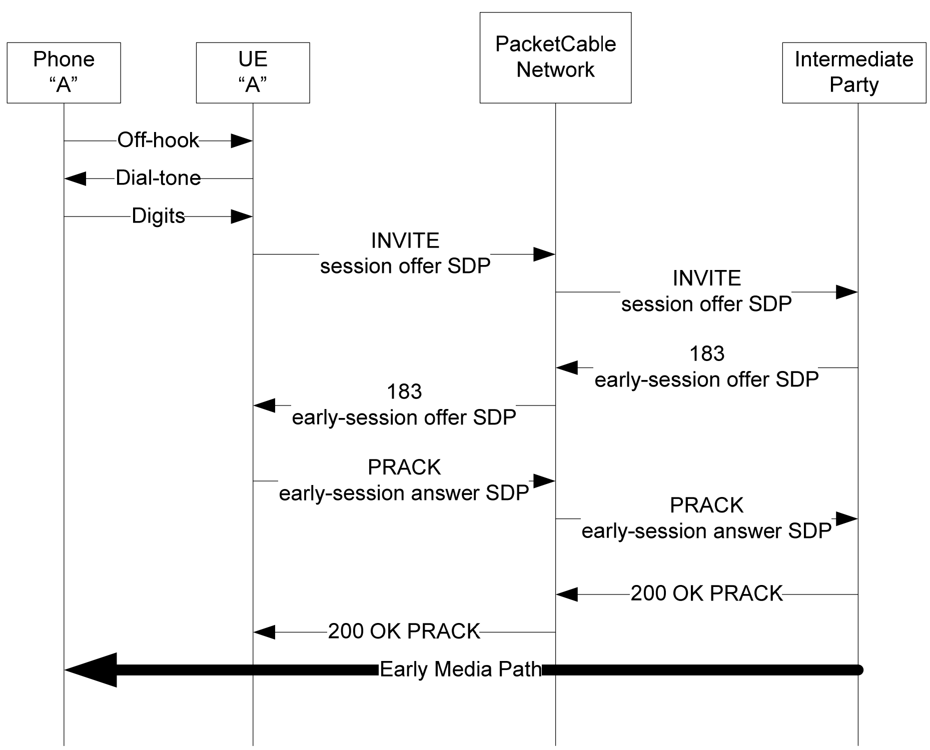

Figure 20 - Early Media from intermediate party

Figure 21 illustrates setup of early media for calls placed from the PSTN to the PacketCable network.

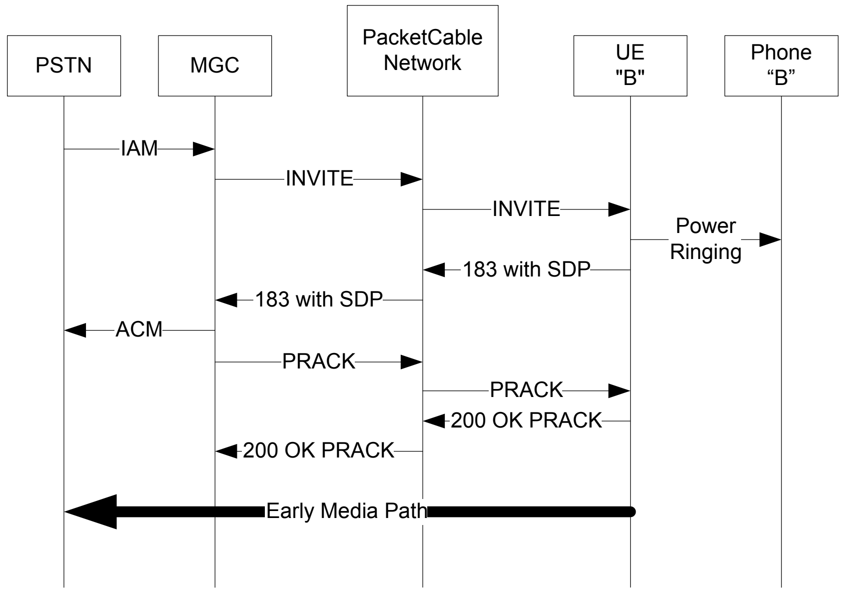

Figure 21 - Early Media; PSTN Origination

7.1.10 No Answer Timeout

7.1.10.1 Feature Description

A dialog invitation may sometimes be not answered: the destination UE continuously applies "Power Ringing" and no end user or device (modem/fax) responds to the ringing signal (no change from "On Hook" to "Off Hook" state on the destination UE). It is undesirable to indefinitely continue in the Power Ring state. Therefore, it is appropriate to terminate an INVITE request which does not result in a 200 OK response (off hook state detection). The No Answer Timeout provides the means to terminate such an unsuccessful dialog initiating request when the timer elapses before the request is answered.

7.1.10.2 Activation & Deactivation

The provisioned timer named 'No Answer Timeout' is used to determine when an INVITE request should be canceled due to the remote communication peer not answering.

If the originating UE goes back to the on hook state at any time during the No Answer Timeout duration, the originating UE MUST send a CANCEL message for the original INVITE.

7.1.10.3 Feature Execution

The destination UE MUST implement the No Answer Timeout timer. The following procedures apply for the processing of No Answer Timeout timer:

- The destination UE MUST start the timer "No Answer Timeout" after sending an 18x response to an initiating request.
- The destination UE MUST reset the No Answer Timeout timer if the destination UE detects an "Off Hook" state.
- If an "Off Hook" state is not detected and the No Answer Timeout timer expires, the destination UE MUST send a 480 Temporarily Unavailable response to the INVITE and stop ringing the phone.

- In response to the expiration of the no-answer timeout, the destination UE MUST disable the Power Ring state.

The following requirements apply to the originating UE:

- In response to 480 Temporarily Unavailable response, the originating UE MUST respond with an ACK.
- After receiving the 480 Temporarily Unavailable response, the originating UE MAY request a Network Announcement per Section 7.1.6 Announcements.

7.1.10.4 Feature Data

Table 11 summarizes the feature data items that are defined to support the No Answer Timeout feature.

Table 11 - No Answer Timeout Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
No Answer timer duration (Note 1)	Integer (seconds)	Non-volatile in XDS, Volatile in UE	Per public user identity	XDS	XDS	UE	Mandatory
Note 1: It is recommended that the value of this timer be set to a value higher than the timer for Call Forwarding Don't Answer so that Call Forwarding feature is invoked before the No Answer timer expires. Note that Call Forwarding Don't Answer feature will CANCEL the INVITE to the destination UE before forwarding the call.							

7.1.11 Response Codes

Table 12 specifies the list of response codes that MUST be supported by UE, AS, P-CSCF, S-CSCF and E-CSCF. These response codes are defined in [RFC 3261]. Individual features specified in this document may also specify actions to response codes that override the response codes specified in Table 12.

Table 12 - Response Codes

Response Code	Action
180 Ringing	If SDP is not specified, the UE MUST apply local ringback tone to the phone. If SDP is specified, the UE MUST play back audio received in the RTP stream.
183 Session Progress	Action is depending on the situation. For early media, it means setting media stream active, for example, a=sendrecv or a=recvonly.
200 OK	The UE MUST sends an ACK.
302 Moved Temporary	The receiving UE MUST use the address in the Contact header to send a new request.
433 Anonymity Disallowed	The receiving UE MUST follow the procedures in Section 7.2.7.
480 Temporarily Unavailable	The receiving UE MUST follow the procedures in Sections 7.1.4.2, 7.1.10, 7.4.3.4, 7.4.3.5, and 7.7.1.6.
486 Busy Here	The UE MUST apply busy tone or announcement to the phone, respond with an ACK and terminate the SIP session.
All Other Response Codes	The UE MUST respond according to [RFC 3261], and all other IETF RFCs required in this specification that defines new response codes according to the IANA SIP registry.

7.1.12 Feature State Synchronization between Application Server and User Equipment

Certain RST features require the synchronization of feature state information between an application server and the user equipment. The features that require this capability are:

- Call Forwarding Variable: for the playing of a special conditions dial tone when the feature is active, and for the playing of a ringsplash whenever a call is forwarded by the application server.
- Do Not Disturb: for the playing of a special conditions dial tone when the feature is active.

For these features, the application server may become aware of a change of state information in the network, and needs a method for notifying user equipment of the change in feature state. To that end, UEs that have either the Call Forwarding Variable or Do Not Disturb feature available MUST send periodic SUBSCRIBEs to the ua-profile Event Package.

When an application server records a state change to the feature data for a particular public identity for the Call Forwarding Variable or Do Not Disturb features, it MUST trigger a NOTIFY message to be sent to the UEs that are subscribed to the ua-profile Event Package for that public identity and feature.

The content of the NOTIFY message bodies for each of these features is defined in the respective feature sections.

7.2 Caller ID

7.2.1 Caller ID Presentation

Several features, described in the following sections, cooperate to provide Caller Identification (Caller ID) functionality. Implementation is distributed among the originating UE, originating P-CSCF, originating Application Server, terminating Application Server, terminating P-CSCF, terminating UE, ingress and egress MGC.

The following is an outline of the process:

- When a UE registers, the S-CSCF receives calling name information from the HSS, and passes it to the P-CSCF.
- When making a call the originating UE may provide a hint about the identity of the user originating the request. It may also indicate preferences regarding the presentation or blocking of the display of the identity information to a recipient outside the provider's trust domain, such as the terminating UE, as specified in [PKT 24.229].
- The originating P-CSCF, using the hints from the UE if available, inserts an assertion about the identity of the originator into the request, as specified in [PKT 24.229]. The assertion contains the calling name if it was received from the S-CSCF.
- In the case of calls originating in the PSTN, an ingress MGC takes identity information from the PSTN request and uses it to insert an identity assertion in the resulting SIP request. It also takes information about the presentation status of the PSTN request and uses it to indicate Caller ID representation status in the SIP request. In some cases it may perform a TCAP query to determine the desired presentation status.
- A terminating Application Server may override the Caller ID blocking status of the request. If calling name delivery is enabled but the calling name is not included in the identity assertion, then the terminating AS may perform a TCAP query to obtain the calling name, and insert it into the identity assertion.
- The terminating S-CSCF or P-CSCF removes the identity assertion from the request before forwarding it if Caller ID blocking is enabled. See [PKT 24.229].
- If the Caller ID display feature is enabled on the terminating UE, and the identity information is present in the incoming SIP request, a terminating UE displays the available identity information to the user.
- In the case of calls terminating in the PSTN, an egress MGC takes the identity information and blocking status from the incoming SIP request and transforms it into presentation status for the resulting PSTN message.

The document [GR 391] defines the Caller ID feature using the notion of Presentation Status (PS). Telcordia uses two presentation status attributes – one for calling number and one for calling name. Each may take the value "public" or "anonymous." These two attributes may be controlled independently and determine whether, respectively, the calling number and calling name (if available) are presented to the callee. If the attribute value is "public" then the corresponding aspect of identity is displayed. If the value is "anonymous" then the identity is not displayed. The presentation status values for a request are derived starting from a pair of Permanent Presentation Status (PPS) values (for number and name) and modifying those based on per-call features invoked by the originating UE.

Within SIP requests, the PacketCable network elements follow the procedures of [PKT 24.229] and assert subscriber identity using the P-Asserted-Identity header. This header contains a URI and an optional display name. The user part of the URI is used to represent the calling number, while the display name represents the calling name. There may be up to two P-Asserted-Identity headers in a SIP request. If only one P-Asserted-Identity is present, it may contain a SIP, SIPS, or TEL URI. If two P-Asserted-Identity fields are present, then [RFC 3325] requires that one must contain a SIP or SIPS URI and the other must contain a TEL URI.

If two P-Asserted-Identity headers are present, the UE SHOULD use the following procedures to select the values of the P-Asserted-Identity header to use for Caller ID display:

- Make a preliminary choice of which P-Asserted-Identity header value to use according to Table 13.
- If the Table does not specify a choice, then if exactly one has a display name, choose that one, else choose the first one.
- If the chosen one has no display name, but the other one does have a display name, then use the URI from the chosen one and add the display name from the other one to it.
- The chosen header value is used as the P-Asserted-Identity header in the procedures of this section and subsections.

If the P-Asserted-Identity header value is required to be modified for the purposes of Caller ID display procedures, the unmodified or original P-Asserted-Identity header MUST be included in subsequent SIP signaling.

Table 13 - P-Asserted-Identity Header Selection

			TEL URI	
			Global-Number	**Local-Number**
SIP/SIPS URI	User=phone	Global #	—	SIP/SIPS
		Local #	TEL	—
	User≠phone	Global #	TEL	SIP/SIPS
		Local #	TEL	—

The PacketCable network elements follow the procedures of [PKT 24.229] and the desired presentation status is specified by the Privacy header as defined in [RFC 3323]. The Privacy header value of 'id' which means to withhold the P-Asserted-Identity header from untrusted elements including the terminating UE is defined in [RFC 3325]. Network elements such as a P-CSCF or I-CSCF follow the trust model defined in [PKT 24.229]. Such network elements interpret the lack of a Privacy header, or a Privacy header that does not contain the value 'id' as meaning that a P-Asserted-Identity header should be retained in the request when it is forwarded by the network element. The Privacy header does not have the capability to independently control the presentation of calling name and calling number. Both are either presented or blocked from the UE. For this reason, PacketCable has only one Presentation Status (PS) attribute per call, and only one Permanent Presentation Status attribute per Public User Identity.

An originating UE uses the P-Preferred-Identity header to indicate which public identity it would like asserted for a particular request. It has the same format as the P-Asserted-Identity header. It may be inserted by the originating UE to express a preference, and then is usually considered by the originating P-CSCF (or sometimes another core element) when choosing what value to include in P-Asserted-Identity. (An originating UE is not permitted to insert a P-Asserted-Identity header.) When the P-Asserted-Identity header is added to the request the P-Preferred-Identity header is removed. The P-Preferred-Identity header is not present during termination processing of the request.

When a call is originated by a UE, the UE is solely responsible for determining the Presentation Status of the call. The UE MUST determine the presentation status for a call by a combination of the Permanent Presentation Status parameter and the per-call Caller ID feature that has been invoked, if any. Table 14, specifies how these are combined.

Table 14 - Effects of Per-Call Blocking Features on Presentation Status

PPS value for the originating subscriber	Per-call Caller ID Blocking feature	Resulting Presentation Status
anonymous	none	anonymous
public	none	public
public	CNDB (see Note 1) and/or CNAB (see Note 2)	Anonymous (see Note 4)
anonymous	CNDB and/or CNAB	Public (see Note 4)
anonymous or public	CIDS Suppression (see Note 3)	anonymous
anonymous or public	CIDS Delivery	public
Note 1 CNDB: Calling Number Delivery Blocking Note 2 CNAB: Calling Name Delivery Blocking Note 3 CIDS: Caller ID Delivery and Suppression Note 4 This handling of CNDB and CNAB is not conforming to Telcordia because each one blocks both number and name. A choice has been made to err on the side of confidentiality – a feature that would block name or number in the PSTN is interpreted as blocking both name and number in PacketCable. An alternative would be to these usages as errors.		

7.2.1.1 Feature Implementation

When the UE originates a call, it MUST determine what presentation status applies for this call, as specified in Table 14, and so indicate in the request.

The presentation status for a call is denoted in the request as follows:

- If the presentation status is "public":

  If the UE has registered with more than one Public Identity, then the UE MUST insert a P-Preferred-Identity header containing the Public User Identity for the line. This changes a behavior that is optional in IMS [PKT 24.229] to mandatory. If there is more than one Public User Identity for the line, one in telephone number format SHOULD be used.

  The UE MUST NOT include a Privacy header containing a value of "id."

  The UE SHOULD include the chosen Public User Identity in the From header.

  The UE SHOULD include the calling name of the chosen Public User Identity as the display name in the From header.

  (While the From header is not used for caller identification by a terminating UE, including it has the potential to enhance interoperability with peers that are not PacketCable compliant).

- If the presentation status is "anonymous":

  The UE MAY insert a P-Preferred-Identity header containing the Public User Identity for the subscriber. If there is more than one Public User Identity for the line, one in telephone number format SHOULD be used.

  The UE MUST include a Privacy header containing a value of "id."

  The UE SHOULD include the URI <sip:anonymous@anonymous.invalid> in the From header.

  The UE SHOULD include the value "anonymous" as the display name in the From header.

  (While the From header is not used for caller identification by a terminating UE, it may be used by a peer that is not PacketCable-compliant. Using "anonymous" rather than the actual identity protects against disclosure of identity that is intended to be blocked.)

The UE SHOULD populate the P-Preferred-Identity header with either a TEL URI, or a SIP URI with a parameter of 'user=phone', otherwise PSTN interoperability may not function correctly.

The actions at the P-CSCF and S-CSCF with regard to handling the received P-Preferred-Identity header are defined by [PKT 24.229].

When the S-CSCF processes a successful registration, the information it obtains from the HSS may contain a calling name for the registered public identity. The S-CSCF conveys the calling name to the P-CSCF in the P-Associated-URI header if the calling name was received from the HSS as a result of a successful registration.

When the P-CSCF processes a successful registration, it saves any display names included in the returned P-Associated-URI parameter, for use in including calling name in asserted identities.

7.2.1.2 Feature Data

Table 15 shows feature data items that are applicable to multiple Caller ID features.

Table 15 - Caller ID Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Public User Identity(s) that may be used by the UE to originate calls	SIP or TEL URI	Volatile on UE	Per UE	HSS	HSS	UE, S-CSCF, P-CSCF	None
Calling Name	String	Non-volatile on HSS	Per public user ID	HSS	HSS	S-CSCF	None
Permanent Presentation Status (PPS)	Enum (public, anonymous)	Volatile on UE, Non-volatile on XDS	Per public user ID	XDS	XDS	UE	Mandatory
Per Call Presentation Status (Public/Anonymous)	Enum (public, anonymous)	Volatile	Per public user ID	UE	UE	UE	None
Per Call Presentation Status Used? (Yes/No)	Boolean	Volatile	Per Public User ID	UE	UE	UE	None

7.2.2 Caller ID Delivery

7.2.2.1 Feature Description

Caller ID Display is a feature that allows a subscriber to see who is calling him before he answers the call. Caller ID Display consists of two sub-features: Calling Number Display (CND) and Calling Name Display (CNAMD).

Except as noted here, Caller ID functionality must match that described in Telcordia documents [GR 31] (CND), [GR 1188] (CNAMD), and [GR 30] (Delivery of data).

The user interface descriptions in [GR 31] must be followed for UEs with analog interfaces. UEs with other interfaces are allowed to provide alternative user interfaces because different types of UEs have widely varying user interface capabilities. In addition to the wide range of UEs, a service provider may offer a web client mechanism to allow a subscriber to manage his or her service data. For this section, however, descriptions are limited to interactions with the telephony client, rather than a service management web client.

In particular for the analog interface, the Caller ID information, consisting of the date and time of the call, plus the calling party's number, and optionally his name, is delivered to the analog interface in the first four seconds of silence after the first ring. The information is transmitted as follows:

> <2 digit month><2 digit day of the month><2 digit hour (24 hour clock)><2 digit minutes<10 digits number><15 character name>

If no information can be obtained, an "O" is sent (originally it meant Out of area). If the caller has blocked the delivery of his information, a "P" is sent (Private).

Caller ID is also delivered with calls that are routed to voice mail systems.

7.2.2.2 Feature Execution

If the Calling Name Display Feature is active, it is recommended that there be a filter in the User Profile that invokes an Application Server for a terminating request when the following conditions are met:

- the request has a P-Asserted-ID with the Display Name absent
- the URI in the P-Asserted-ID header contains an E.164 number
- the request contains no Privacy header, or a Privacy header is present and contains the token "none," or a Privacy header is present that does not contain either the token "id" or the token "header."

The AS MUST query a CNAM DB and update the display name in the P-Asserted-ID with either the name from the DB, or a display name of "" if no value can be obtained from the CNAM DB. (The AS may use TCAP, ENUM, or any other mechanism to perform the query.)

If the Calling Name Display feature is absent or inactive, there may be a filter in the User Profile that invokes an Application Server for a terminating request when:

- there is no Privacy header, or
- there is a Privacy header containing neither the token "none" nor the token "id".

In this case, the AS MAY insert the token "id" into an existing Privacy header or insert a new Privacy header containing the token "id".

Note: If there is a Privacy header containing the token "none," the AS is forbidden by [RFC 3323] from removing the "none" token. In this case, if the AS wishes to prevent the Caller ID from being made available to the UE, the AS MUST prevent the request from being delivered to the UE.

For calls placed to a public identity that has subscribed to Caller ID Display, the called party's P-CSCF will examine the Privacy header in the incoming INVITE and forward or strip the P-Asserted-Identity header(s) to the UE according to the rules that govern transmission of identity information to an un-trusted domain/entity (the UE).

A UE displays the calling number if and only if the CND feature is activated, and displays the calling name if and only if the CNAMD feature is activated. The following steps specify how the calling number and name is displayed.

For a non-analog UE, the UE MAY display the Caller ID Display information from the P-Asserted-Identity header(s) in an implementation dependent manner. However, the UE SHOULD display at least the date/time of the incoming call, the name, and, if present, the telephone number of the caller.

For the analog UE, the UE MUST display the Caller ID Display information in accordance with Telcordia [GR 30]. The date and time of the call is obtained from the UE clock.

In order to preserve compatibility with the large number of existing analog Caller ID Display units already in existence in North America, the analog UE MUST conform to the following requirements for creating the Caller ID Display data from the information in the INVITE:

1. If no P-Asserted-Identity header is present, then a "P" is transmitted in the number field, and nothing in the name field.
2. If a SIP or SIPS URI is present in the P-Asserted-Identity header, then the first 15 characters of the display name, if present, are transmitted in the name field. Otherwise, nothing is transmitted in the name field.
3. If a TEL URI is present in the P-Asserted-Identity header and contains a local-number as defined in [RFC 3966] (if the first character is NOT a "+"), then the local-number-digits component is extracted and all visual-separator characters removed. Then all the remaining characters up to the first 10 are transmitted in the number field.
4. If a TEL URI is present in the P-Asserted-Identity header and contains a global-number as defined in [RFC 3966] (if the first character is a "+") then the global-number-digits component is extracted and all visual-separator characters as well as the leading "+" character removed. If the initial characters in the result match the default country code, then they MAY be stripped out. Then all the remaining characters up to the first 10 SHOULD be transmitted in the number field.
5. If no TEL URI is present in the P-Asserted-Identity header, but there is a SIP or SIPS URI with a user part beginning with "+," the user part is used to determine the number field in accordance with the rules in the preceding step.
6. If none of the steps above cause the number field to be transmitted, then an "O" is transmitted in the number field.

7.2.2.3 Feature Data

Table 16 shows feature data items that are defined to support the Caller ID display feature.

Table 16 - Caller ID Display Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
CND feature activation status	Boolean	Volatile on UE, Non-volatile on XDS	Per public user ID	XDS	XDS	UE	Mandatory
CNAMD feature activation status	Boolean	Volatile on UE, Non-volatile on XDS	Per public user ID	XDS	XDS	UE	Mandatory
Current Time of Day – location invariant	UTC	Volatile	Per network	UE	UE	UE	None

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Adjustment from location invariant time to time at current location	Time delta (minutes)	Volatile on UE, Non-volatile on XDS	Per UE	XDS	XDS	UE	Mandatory
Default Country Code	String	Volatile on UE, Non-volatile on XDS	Per public user ID	XDS	XDS	UE	Mandatory

7.2.2.4 Feature Interactions

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

- Specifications of presentation apply, but specifications of the mechanisms used to obtain the information displayed do not apply.
- AC/AR subsection does not apply directly. Multiparty Lines subsection has limited applicability because a UE cannot be a multiparty line.
- CND subsection of [GR 1188] applies except that usage sensitive CND and CNAMD do not apply.
- For interactions with AC and AR, see Section 7.7.2 and Section 7.7.1.

7.2.3 Caller ID Per-Line Blocking

7.2.3.1 Feature Description

Caller ID Per-Line Blocking is a feature that allows configuration of a default Permanent Presentation Status (PPS) value of "public" or "anonymous" for the presentation of Caller ID on a per-line (Public User ID) basis. This is similar to the CIDB feature defined in [GR 391]. CIDB permits specifying PPS of "public" or "anonymous" separately for calling number and calling name. The feature described here defines a single default value that applies to both calling number and calling name. This feature is not activated or deactivated through the use of feature codes – it is activated and deactivated only via provisioning.

When a caller makes a call and does not invoke the per-call Caller ID blocking features, then the Permanent Presentation Status value defined here applies to both the calling name and calling number of the call.

When a caller makes a call that invokes per-call Caller ID blocking features, those features interact with the provisioned PPS to determine the Caller ID presentation attributes of the call. The mechanics of invocation are specified in the sections pertaining to those features.

7.2.3.2 Feature Implementation

Caller ID Per-Line Blocking is carried out by the calling UE. The UE MUST be provisioned with read-only versions of the PPS value for each line it supports, as well as the Public User Identity for that line. The UE MAY also be provisioned with the Calling Name value for that line.

The application of Caller ID Per-Line Blocking to a call is specified in Section 7.2.4.

7.2.3.3 Feature Interactions

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

- [GR 391] describes multiple features. Only the interactions pertaining to per-line name and number blocking apply here.
- For PacketCable there is only one PPS (Permanent Presentation Status) value governing the presentation of both name and number. References in [GR 391] to the PPS for either name or number refer to this single value.
- Specifications of presentation apply, but specifications of the mechanisms used to obtain the information displayed do not apply.
- The following [GR 391] interactions do not apply: Basic Business Group, Bulk Calling Line Identification, Charge-a-call, Coin Service, Multi Party Lines of more than two parties, Multi-Switch Business Groups, Private Branch Exchange, Remote Switching Units and Two-Party Lines.
- The interactions with CND and CNAMD are specified in Section 7.2.2 of this document.

7.2.4 Caller ID Per-Call Blocking (CIDS-Suppression)

7.2.4.1 Feature Description

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

- Goes off-hook and received dial tone.
- Enters the assigned vertical feature code. The feature code is provisioned in the UE. (The suggested default value is *67.)
- Receives dial tone, or a confirmation tone followed by a recall dial tone.
- Dials the telephone number of the called party.

For this one call, the permanent presentation status value is overridden with the value "anonymous" as specified in 7.2. If the caller goes on hook at any point prior to completion of dialing the called party number, the feature has no effect.

As a minimum implementation, all tones and messages defined by [GR 391] for support of this feature are required. Optional tone and message sets may be offered as value-added customizable features as long as the required set is part of the implementation.

7.2.4.2 Feature Implementation

CIDS-S is carried out by the calling UE. As part of its CID-SUPPRESS digit map action, the UE MUST check to see if the CID Suppression feature is available; and if the feature is available, set the local privacy state on the UE to suppressed. The UE MUST then cause recall dial tone to be played, and be ready to accept dialed digits. If the CID Suppression feature is not available, then the UE MUST cause an error tone or message to be played.

7.2.4.3 Feature Data

Table 17 summarizes the feature data items that are defined to support the CIDS-S feature.

Table 17 - Caller ID Per-Call Blocking Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Feature availability status	Boolean	Volatile on UE, Non-volatile at XDS	Per public user ID	XDS	XDS	UE	Mandatory
Confirmation Tone after vertical feature code	Tone	Volatile on UE, Non-volatile on XDS	Per Action	XDS	XDS	UE	Mandatory
Error Tone after vertical feature code failure	Tone	Volatile on UE, Non-volatile on XDS	Per Action	XDS	XDS	UE	Mandatory

7.2.4.4 Feature Interactions

ACR

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

AC

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

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

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

AR

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

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

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

Calling Number Delivery

If a caller activates CIDS-S then the callee will not receive Caller ID, including calling number. Detailed results are covered in Section 7.2.2.

Calling Name Delivery

If a caller activates CIDS-S then the callee will not receive Caller ID, including calling name. Detailed results are covered in Section 7.2.2.

Operator Services

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

Screening List Services

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

Speed Dialing

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

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

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

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

Three-Way Calling

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

7.2.5 Caller ID Per-Call Delivery (CIDS-Delivery)

7.2.5.1 Feature Description

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

- Goes off-hook and received dial tone.
- Enters the assigned vertical feature code. The feature code is provisioned in the UE. (The suggested default value is *82.)
- Receives dial tone, or a confirmation tone followed by a recall dial tone.
- Dials the telephone number of the called party.

For this one call, the permanent presentation status value is overridden with the value Public, as specified in 7.2. If the caller goes on hook at any point prior to completion of dialing the called party number the feature has no effect.

As a minimum implementation, all tones and messages defined by [GR 391] for support of this feature are required. Optional tone and message sets may be offered as value-added customizable features as long as the required set is part of the implementation.

The deviations from [GR 391] are the same as specified for CIDS-S.

7.2.5.2 Feature Implementation

CIDS-D is carried out by the calling UE. As part of its CID-DELIVER digit map action, the UE MUST check to see if the CID Delivery feature is available, and if the feature is available, set the local privacy state on the UE to display. The UE MUST then play recall dial tone, and be ready to accept dialed digits. If the CID Delivery feature is not available, the UE MUST play an error tone or message.

7.2.5.3 Feature Data

Table 18, summarizes the feature data items that are defined to support the Caller ID Per-Call Delivery feature.

Table 18 - Caller ID Per-Call Delivery Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Feature availability status	Boolean	Volatile on UE, Non-volatile on XDS	Per public user ID	XDS	XDS	UE	Mandatory
Confirmation Tone after vertical feature code	Tone	Volatile on UE, Non-volatile on XDS	Per Action	XDS	XDS	UE	Mandatory
Error Tone after vertical feature code failure	Tone	Volatile on UE, Non-volatile on XDS	Per Action	XDS	XDS	UE	Mandatory

7.2.5.4 Feature Interactions

AC

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

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

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

AR

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

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

An AR customer may use this feature on the last call that the customer received regardless of the caller's presentation statuses. However, if the call that the AR customer received was originally marked "anonymous", the

SPCS does not disclose the Caller ID to the AR customer when the call is finally established and the AR customer gets ring back.

Calling Number Delivery

If a caller activates CIDS-D then the callee will receive Caller ID, including calling number, regardless of the permanent presentation status. Detailed results are covered in Section 7.2

Calling Name Delivery

If a caller activates CIDS-D then the callee will receive Caller ID, including calling name, regardless of the permanent presentation status. Detailed results are covered in Section 7.2.

Operator Services

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

Screening List Services

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

Speed Dialing

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

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

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

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

Three-Way Calling

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

7.2.6 Caller ID with Call Waiting (CIDCW)

CIDCW is not a separate feature. See Section 7.5.2.

7.2.7 Anonymous Call Rejection

7.2.7.1 Feature Description

The Anonymous Call Rejection (ACR) service allows the user to reject incoming calls where the identity of the originating user is or has been restricted.

When active the service rejects originator identity restricted calls without alerting the PacketCable subscriber. The originating user is provided with an appropriate indication that the call has been rejected by an instance of an ACR service.

The behavior of the service is as described in Telcordia [GR 567] the requirements of which must be met by a PacketCable network unless otherwise specified here. The following sections and requirements within normative reference [GR 567] do not apply to PacketCable networks. These deviations are additional to any divergence from [GR 567] detailed elsewhere in this document:

- The billing requirements found in [GR 567] do not apply.
- All requirements in [GR 567] related to business groups do not apply.
- The traffic and maintenance measurements requirements in [GR 567] do not apply.
- The maintenance requirements in [GR 567] do not apply.

7.2.7.2 Feature Activation and Deactivation

The availability of ACR to a subscriber is via prior arrangement between the subscriber and the PacketCable network provider. Once available and enabled the user activates and deactivates the service by dialing of vertical service codes (VSCs). These are typically *77 for activation and *87 for deactivation. Alternately, a service provider, as part of their service offering, could provide subscriber access to a web portal that supports ACR activation and deactivation.

On activation of ACR, as part of its ACR-ACTIVATE digit map action, if the ACR feature is available, the UE MUST send an INVITE with the request URI capturing the VSC as per the network requirements. On entry of the ACR de-activation code, as part of its ACR-DEACTIVATE digit map action, if the ACR feature is available, the UE MUST send an INVITE with the request URI capturing the VSC as per the network requirements. If the ACR feature is not available, the UE MUST apply recall dial-tone to the phone and take no further action.

On successful activation or deactivation of the feature the ACR AS MUST provide a confirmation tone or announcement to the subscriber. If an error occurs in the activation or deactivation of ACR the ACR AS MUST provide an error tone or appropriate announcement.

7.2.7.3 Feature Execution

The execution of the ACR service is provided by the ACR Application Server (AS) in the terminating (ACR subscriber's) network.

It is assumed that when an incoming SIP INVITE is received by the terminating S-CSCF it is proxied to the ACR AS by the S-CSCF if the filtering logic of the called UE's service profile dictates to do so.

If the ACR feature is not provisioned for the terminating subscriber or has been de-activated then the ACR AS proxies the INVITE back to the S-CSCF.

7.2.7.3.1 Determination of Restricted Identity

The ACR AS determines based on the contents of the INVITE whether the service needs to be applied. The originating identity of the caller is determined to be restricted if the following conditions hold:

- A Privacy header indicating a privacy level of "id" is present in the INVITE, or
- A Privacy header indicating a privacy level of "user" is present in the INVITE, or
- A Privacy header indicating a privacy level of "header" is present in the INVITE,

and

- The caller is not an authorized user with ACR override privileges. Calls from Emergency Services and Operator Services personnel are not blocked by the ACR feature, since these calls are not anonymous, and has reserved values in their P-Asserted-Identity fields.

7.2.7.3.2 Actions when Identity is Restricted

If originating identity is restricted, the ACR AS MUST perform the following actions:

- Provide an appropriate audible indication to the originator that the call is being rejected due to the application of ACR.
- Reject the INVITE by sending a 433 Anonymity Disallowed final response as defined by reference [ID SIP-ACR].

If the originating identity is restricted, the ACR AS SHOULD include a Warning header in the final response with a "warn code" of 399 and "warn text" as determined by the service provider.

7.2.7.4 Feature Data

Table 19 summarizes the feature data items that are defined to support the ACR feature.

Table 19 - ACR Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Feature Availability Status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
Feature Activation Status	Boolean	Non-v volatile	Per public user ID	AS	AS	AS	None
Caller Treatment or Announcement Resource (see Note 1)	URI for media server	Non-volatile	Per network	AS	AS	AS	None
Note 1: Not required if the announcement resource is integrated with the AS.							

7.2.7.5 Feature Interactions

The ACR service has the following feature interactions.

Call Waiting

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

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

Caller ID Per-Line Blocking

If the originating user has activated CNDB or CNAB and that user is not an authorized user with ACR override, then all calls from that user are rejected by a subscriber with ACR.

Call Forwarding

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

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

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

Automatic Callback (AC)

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

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

Automatic Recall (AR)

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

Distinctive Ringing/Call Waiting (DRCW)

If a subscriber has both ACR and DRCW, then DRCW takes precedence over ACR. That is, if the restricted identity is that of an originator that is configured by the ACR subscriber to trigger DRCW, then ACR processing is not invoked.

Emergency Services

Emergency call back overrides Anonymous Call Rejection.

Operator Services Emergency Interrupt overrides Anonymous Call Rejection.

Solicitor Blocking

Anonymous Call Rejection takes precedence over the screening by keypress version of Solicitor Blocking.

The screening by greeting record version of Solicitor Blocking takes precedence over Anonymous Call Rejection.

Do Not Disturb

Do Not Disturb takes precedence over Anonymous Call Rejection

7.2.7.6 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence. Figure 22 illustrates the call flow for ACR activation or deactivation.

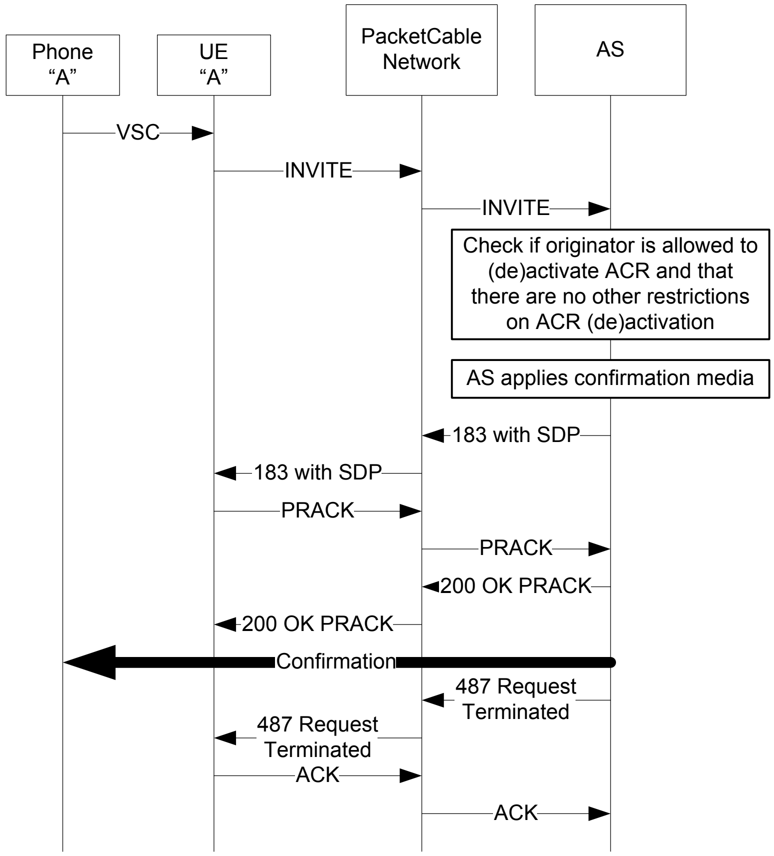

Figure 22 - Anonymous Caller Rejection (ACR); Activation/deactivation

Figure 23 illustrates the call flow for the case where the terminating subscriber (B) has activated ACR and the originator (A) has a restricted identity.

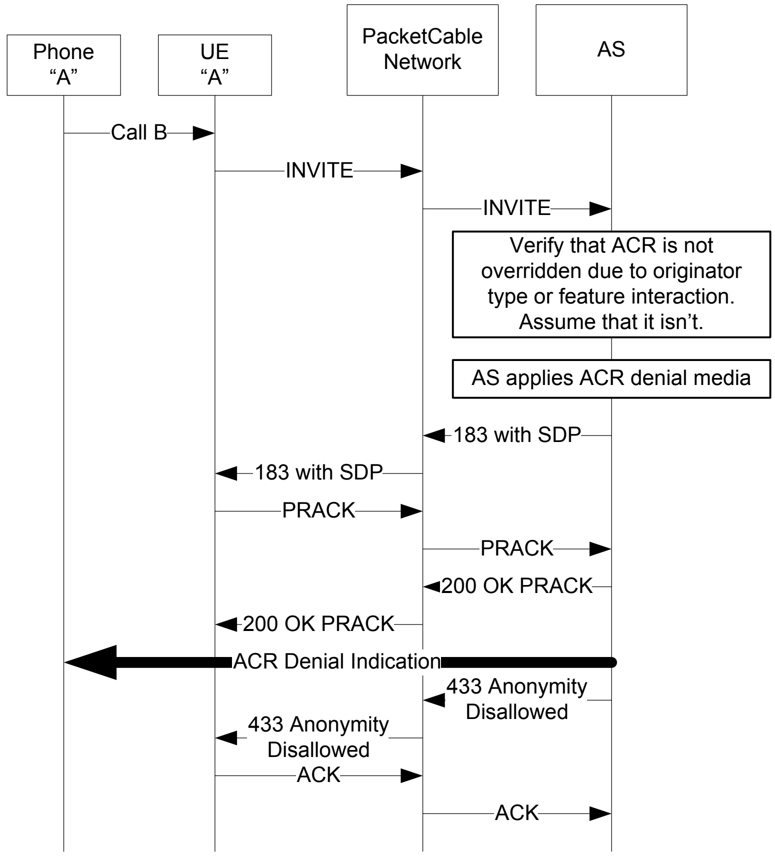

Figure 23 - ACR; Originator has a restricted identity

7.2.8 MGC Considerations

The MGC requirements and procedures (including SIP to ISUP interworking) for supporting Caller Id for calls between PacketCable and PSTN are specified in [CMSS1.5].

7.3 Call Forwarding

7.3.1 Overview

There are several different variations of call forwarding. UEs that implement any of the forwarding features described in this section, MUST comply with requirements in Telcordia [GR 586], except as explicitly noted in this document.

PacketCable call forwarding features include Call Forwarding Variable (CFV), Call Forwarding Busy Line (CFBL), Call Forwarding Don't Answer (CFDA), Remote Activation of Call Forwarding (RACF), and Selective Call Forwarding (SCF). These features operate on a per-public identity basis. This means that forwarding affects the public identity, regardless of how many UEs are registered behind that public identity.

Call forwarding functions whether or not the subscriber has a UE registered with the PacketCable network. Therefore, most aspects of Call Forwarding features are provided by an Application Server (AS), which is called the Call Forwarding Application Server (CF AS). For all of the call forwarding features, the original caller's identity is preserved in the P-Asserted-Identity header when forwarding a call.

The CF feature does not require the CF AS to record-route itself into the signaling path. If a CF AS does record-route itself into the signaling path for this feature, all signaling is unnecessarily routed through CF AS, which is inefficient.

7.3.1.1 Preventing Forwarding Loops and Limiting the Number of Forwarding Attempts

For each of the Call Forwarding features, the PacketCable network provides a mechanism to prevent forwarding loops. A call forwarding loop is the scenario that occurs when a targeted subscriber for a call forwards the call to another destination; if the forwarded-to destination also has call forwarding configured, the call can forward back (directly or indirectly) to the original targeted subscriber. When a loop is detected, the network node that performs the detection performs a configurable action, the default being call rejection. A CF AS MUST detect call forwarding loops.

The CF AS MUST support a configurable limit on the number of times an individual call may be subject to forwarding. If the number of forwarding attempts for a single call exceeds this limit, the CF AS MUST perform a configurable action, the default being call rejection.

The CF AS SHOULD support loop prevention and forwarding limit detection via the mechanisms described in this section. However, this mechanism alone may not be sufficient to detect loops when calls are forwarded to networks not supporting these mechanisms (for example, the PSTN or a network not supporting the required SIP headers). Therefore, a CF AS MAY support additional loop prevention and forwarding limit detection methods as long as the requirements of forwarding limit restriction and loop detection are met.

If the CF AS supports the prevention of forwarding loops via analysis of the History-Info header present in the INVITE, then it MUST compare the forward-to address with the set of targeted-to URI (hi-targeted-to-uri) entries from the History-Info header. If there is a match, then a loop has occurred. If no History-Info header is present, then it is not possible to perform loop detection via this mechanism.

If the CF AS determines that a loop has occurred (regardless of the loop detection method used), the CF AS MUST handle the call based upon a configurable action. The CF AS MUST support the default loop detection action of call rejection. The CF AS MUST send a final response appropriate to the type of call forwarding being performed if the call is to be rejected. If the forwarding action is CFBL and the call is to be rejected, the CF AS MUST respond with a 486 User Busy message. If the forwarding action is CFDA and the call is to be rejected, the CF AS MUST respond with a 408 Request Timeout message. If the forwarding action is CFV or SCF and the call is to be rejected, the CF AS MUST respond with a 480 Temporarily Unavailable message.

If the CF AS supports the prevention of forwarding loops by enforcing a maximum number of forwarding attempts, then it MUST calculate the number of forwarding attempts by counting the number of entries in the History-Info header that have nested Reason headers that include protocol-cause parameters and reason-text parameters populated as defined in Section 7.3.1.2. If no History-Info header is present, then it is not possible to determine the number of forwarding attempts via this mechanism.

If the number of forwarding attempts exceeds the configured limit, then the CF AS MUST handle the call based on a configurable action. The CF AS MUST support the default action of call rejection. The CF AS MUST send a final response appropriate to the type of call forwarding performed if the call is to be rejected. If the forwarding action is CFBL and the call is to be rejected, the CF AS MUST respond with a 486 User Busy message. If the forwarding action is CFDA and the call is to be rejected, the CF AS MUST respond with a 408 Request Timeout message. If the forwarding action is CFV or SCF and the call is to be rejected, the CF AS MUST respond with a 480 Temporarily Unavailable message.

7.3.1.2 Setting of the Call Forwarding Parameters

After checking the number of forwards, a number of fields of the INVITE request are set by the CF AS as defined in the following subsections.

7.3.1.2.1 First Forwarded INVITE

The absence of a History-Info header in the INVITE, or the presence of a History-Info with no entries containing a nested Reason header with a protocol-cause parameter, means that this is the first instance of a forwarding action.

When this is the first forward the INVITE has undergone, the following requirements apply. The CF AS MUST set the Request URI to the public user identity where the INVITE is to be forwarded. The CF AS MUST NOT change the contents of the P-Asserted-Identity header in the INVITE.

The CF AS MUST set the History-Info header (redirection information, redirecting number, original called number). Two History-Info entries MUST be generated as described below.

The first added entry in the History-Info header by the CF AS MUST include, as the hi-targeted-to-uri, the URI of the called party that was addressed with this INVITE. If the called party's presentation status is set to anonymous, the CF AS MUST escape the privacy header "history" within the hi-targeted-to-uri; a Reason is not added. If no History-Info header was previously present in the INVITE, then the CF AS MUST set the Index to index=1. If this is an additional entry to an already present History-Info header, then the CF AS MUST set the index according to the rules in [RFC 4244].

The second added entry in the History-Info header by the CF AS MUST include, as the hi-targeted-to-uri, the address to where the INVITE is forwarded. If no History-Info header was present prior to this procedure, then the CF AS MUST set the index to index=1.1. If this is an additional entry to an already present History-Info header, then the CF AS MUST set the index according to the rules in [RFC 4244].

When adding a second entry, the CF AS MUST also include a nested Reason header (redirecting reason and redirecting indicator) escaped in the History-Info header, this is populated according to the forwarding conditions.

- [RFC 4458] defines the mapping between the forwarding conditions and the coding of the protocol-cause parameter in the Reason header. The CF AS MUST populate the Reason header with a protocol-cause value of "486" and a reason-text value of "CFBL" when the forwarding condition is CFBL. The CF AS MUST populate the Reason header with a protocol-cause value of "408" and a reason-text value of "CFDA" when the forwarding condition is CFDA. The CF AS MUST populate the Reason header with a protocol-cause value of "302" and a reason-text value of "CFV/SCF" when the forwarding condition is CFV or SCF.
- Finally, the CF AS MUST set the To header as per the following rules. If the forwarding party does not request privacy to be applied (that is, the presentation status is set to public), then the CF AS MUST NOT change the To header. When the forwarding party requests privacy (that is, the presentation status is set to anonymous), the CF AS MUST change the To header to be the URI to where the INVITE is forwarded.

7.3.1.2.2 Second or Subsequent Forwarded INVITE

When this is the second or subsequent forwarding of the INVITE, the CF AS MUST add a new entry to the History-Info header according to the rules defined in [RFC 4244]. If the history entry representing the forwarding party already contains the correct privacy value for the forwarding party (in an escaped privacy header), then the CF AS MUST NOT modify the History-Info header. Otherwise, if the forwarding party requests privacy (that is, the presentation status is set to anonymous), the CF AS MUST ensure the privacy header "history" is escaped within the hi-targeted-to-uri.

The entry MUST contain, as the hi-targeted-to-uri, the address to where the INVITE is forwarded. The CF AS MUST populate the Reason header (redirecting reason and redirecting indicator) escaped in this history-info header according to the forwarding conditions and notification subscription option.

The CF AS MUST code the protocol-cause parameter in the Reason header based on forwarding conditions as defined in [RFC 4458]. The CF AS MUST populate the Reason header with a protocol-cause value of "486" and a reason-text value of "CFBL" if the forwarding condition is CFBL. The CF AS MUST populate the Reason header with a protocol-cause value of "408" and a reason-text value of "CFDA" if the forwarding condition is CFDA. The CF AS MUST populate the Reason header with a protocol-cause value of "302" and a reason-text value of "CFV/SCF" if the forwarding condition is CFV or SCF. The CF AS MUST set the Request URI to the public user identity where the INVITE is to be forwarded.

The CF AS MUST NOT change the contents of the P-Asserted-Identity header in the INVITE.

The CF AS MUST NOT change the To header field (original destination number) if the forwarding party does not request privacy (that is, the presentation status is set to public). The CF AS MUST change the To header field to be the URI where the INVITE is forwarded if the forwarding party requests privacy (that is, the presentation status is set to anonymous).

Figure 24 illustrates a multiple forwarding scenario, and Table 20 describes how the headers are populated. In this scenario, Alice calls Bob and is forwarded to Charlie, and is then forwarded to Ed and is, in turn, forwarded to Bob.

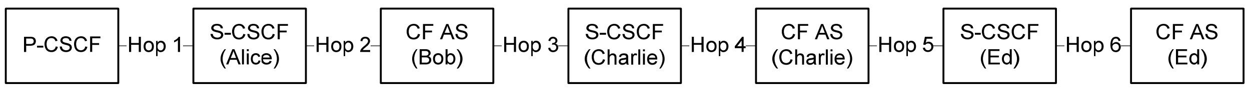

Figure 24 - Multiple Forwarding Scenario

Table 20 - Parameters And Header Fields that are Modified In A CF AS

Element	**P-CSCF**	**S-CSCF (Alice)**
H-I Index Added	1	1.1
Entry Added	Bob@domain	Bob@domain
H-I Header	Bob@domain; index=1	Bob@domain; index=1, Bob@domain; index=1.1
Element	**CF AS (Bob)**	**S-CSCF (Charlie)**
H-I Index Added	1.2,1.2.1	1.2.1.1
Entry Added	Bob@domain, Charlie@domain?Reason=SIP;protocol-cause=302;reason-text="CFV/SCF"	Charlie@domain
H-I Header	Bob@domain; index=1, Bob@domain; index=1.1, Bob@domain; index=1.2, <Charlie@domain?Reason=SIP;protocol-cause=302;reason-text="CFV/SCF">; index=1.2.1	Bob@domain; index=1, Bob@domain; index=1.1, Bob@domain; index=1.2, <Charlie@domain?Reason=SIP;protocol-cause=302;reason-text="CFV/SCF">; index=1.2.1, Charlie@domain; index=1.2.1.1
Element	**CF AS (Charlie)**	**S-CSCF (Ed)**
H-I Index Added	1.2.1.2	1.2.1.2.1
Entry Added	Ed@domain?Reason=SIP;protocol-cause=302;reason-text="CFV/SCF"	Ed@domain
H-I Header	Bob@domain; index=1, Bob@domain; index=1.1, Bob@domain; index=1.2, <Charlie@domain?Reason=SIP;protocol-cause=302;reason-text="CFV/SCF">; index=1.2.1, Charlie@domain; index=1.2.1.1, <Ed@domain?Reason=SIP;protocol-cause=302;reason-text="CFV/SCF">; index=1.2.1.2	Bob@domain; index=1, Bob@domain; index=1.1, Bob@domain; index=1.2, <Charlie@domain?Reason=SIP;protocol-cause=302;reason-text="CFV/SCF">; index=1.2.1, Charlie@domain; index=1.2.1.1, <Ed@domain?Reason=SIP;protocol-cause=302;reason-text="CFV/SCF">; index=1.2.1.2, Ed@domain; index=1.2.1.2.1
Element	**CF AS (Ed)**	
H-I Index Added	No entry is added since loop detection at Ed's CF AS detects a Call Forwarding loop. The forwarding target Bob@domain is already present in the set of URIs contained in the received H-I header.	
Entry Added	None as loop detection detects CF loop	
H-I Header	None as loop detection detects CF loop	

7.3.1.3 MGC Inter-working

To prevent forwarding loops occurring between the PacketCable network and the PSTN, forwarding information needs to be maintained by the MGC when mapping from SIP History Info header to ISUP and vice versa. The MGC requirements and procedures (including SIP to ISUP interworking) for support of forwarding loop detection are specified in [CMSS1.5].

7.3.2 Call Forwarding Variable

7.3.2.1 Feature Description

Call Forwarding Variable (CFV) is a feature that allows a subscriber to activate forwarding of all calls that go to the subscriber's public identity to another location. The forward-to address can be provided through subscriber interaction with the CFV feature, or the network operator can configure the forward-to address for a subscriber. With CFV, the forwarding happens before the called party's UE alerts (rings) the called party, so the forwarded public identity has no opportunity to answer the call.

The behavior of the CFV service, as defined in [GR 580], MUST be met by UEs with analog interfaces with the following exceptions. The following sections and requirements within normative reference [GR 586] do not apply to PacketCable networks. These deviations are additional to any divergences from [GR 586] detailed elsewhere in this document:

- Tones may be substituted for the announcements described in [GR 580].
- Restriction for forwarding to "free numbers" found in [GR 580] do not apply.
- Charging requirements in [GR 580] do not apply.
- Billing requirements in [GR 580] do not apply.
- [GR 580] requirements related to business groups do not apply.
- Traffic and maintenance measurements requirements in [GR 580] do not apply.
- The maintenance requirements in [GR 580] do not apply.

UEs with non-analog interfaces are allowed to provide alternative user interfaces because different types of UEs have widely varying user interface capabilities. In addition to the wide range of UEs, a service provider may offer a web client mechanism the capability of allowing a subscriber to manage his or her service data. For this section, however, descriptions are limited to interactions with the analog interface telephony client, rather than a service management web client.

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

7.3.2.2 Feature Activation and Deactivation

A user activates or deactivates CFV by dialing a vertical service code (VSC) according to the procedures described in Section 7.1.2.

The user can activate or deactivate the CFV feature via the telephony user interface (that is, the phone keypad) by entering a VSC. The CFV VSCs are defined for the UE as part of the Digit Map as defined in Section 7.1.1. Upon matching the user-entered digits with the CFV VSC, the UE carries out the CFV-PROGRAM action. As part of the CFV-PROGRAM action, the UE MUST apply recall dial tone, collect the forward-to address, and send the VSC, along with the collected forward-to address, to the CFV AS via the mechanism specified in Section 7.1.2.

Upon receipt of an INVITE with the CFV activation code, the CFV AS MUST respond with a 200 OK. When the CFV AS receives the ACK from the UE, it MUST play a confirmation tone to the user. If the UE is subscribed to the UA-Profile Event Package, the CFV AS MUST send a NOTIFY to the UE informing it that CFV is activated, per Table 22. When the CFV AS receives the 200 OK to the NOTIFY, it MUST send a BYE to the UE to end the session. If the activation fails, the CFV AS MUST play an error tone or announcement, followed by one second of silence, and then send a BYE. Upon receipt of the BYE, the UE MUST apply the dial tone and be ready to accept dialed digits.

Once call forwarding programming is successful, the CFV AS MUST store the forwarded-to number in such a manner that it remains persistent even in the case of power loss in the UE or network.

If the user-programmable CFV is currently activated, the subscriber tries to activate it to the same number, and verification calling is enabled, the CFV AS MUST treat the attempt just like a second attempt and play a confirmation tone (or an announcement) to the subscriber without actually calling the number.

If CFV verification is enabled, an initial attempt to program CFV MUST result in the CFV AS forwarding the call to the forward-to address. If the call is answered, the CFV AS MUST consider CFV activated. If the call is not answered, and the second verification call is disabled, the CFV AS MUST consider CFV activated. If the call is not answered and the second verification call is enabled, the CFV AS MUST start a two minute timer when the subscriber hangs up on the activation attempt. During the two minutes, the CFV AS MUST store the forward-to address, but CFV is not yet activated.

If CFV second verification call is enabled, and a repeat attempt is made to activate CFV with the same forward-to address and during the two-minute interval, the CFV AS MUST NOT attempt to call the forward-to number. Instead, the CFV AS MUST consider CFV active and play a confirmation tone (or an announcement) to the subscriber. If an attempt is made to activate CFV to a different forward-to address when a two minute timer is running from a previous initial attempt, the CFV AS MUST cancel the two minute timer and treat the new activation attempt as an initial attempt. If a second attempt is made to activate CFV and CFV is not active, but the attempt falls outside the two minute interval, the CFV AS MUST treat the attempt as an initial attempt.

Upon receipt of an INVITE with the CFV de-activation code, the CFV AS MUST respond with a 200 OK. When the CFV AS receives the ACK from the UE, it MUST play a confirmation tone to the user regardless of whether CFV is active or not. If the UE is subscribed to the UA-Profile Event Package, the CFV AS MUST send a NOTIFY to the UE informing it that CFV is de-activated, per Table 22. When the CFV AS receives the 200 OK to the NOTIFY, it MUST send a BYE to the UE to end the session.

If the CFV forward-to number is provided by subscriber, the CFV AS MUST remove the number from its subscriber data on successful deactivation of CFV. If the forward-to number is provided by service provider, CFV is marked inactive, but the CFV AS MUST NOT erase the forward-to number. If there is a two minute timer running for a verification second attempt when the CFV deactivation is received, the CFV AS MUST cancel the two minute timer. CFV deactivation only results in error tone (or announcement) in the unlikely event that CFV is active and cannot be deactivated.

7.3.2.3 Feature Execution

At boot time, if the UE feature availability status indicates support of the CFV feature, the UE MUST send a SUBSCRIBE for the ua-profile event package according to the procedures in [RST PACM].

Upon receipt of a NOTIFY indicating that CFV is active and the UE is provisioned with the Call Forward ring reminder, the UE MUST send a SUBSCRIBE to the CFV AS for the ua-profile event package call forward indication, as described in Table 22, for the call forwarding subscription duration. If, at the expiry of the subscription, CFV is still active, the UE MUST re-subscribe by sending a SUBSCRIBE to the CFV AS for the ua-profile event package call forward indication, as described in Table 22, for the call forwarding subscription duration.

If the special conditions dial-tone is enabled for subscribers with the CFV feature activated, then each time a subscriber with CFV activated goes off hook, the UE MUST play the special conditions dial-tone as a reminder that CFV is active.

Upon receipt of an INVITE destine to a UE for which CFV is active, the CFV AS determines whether the call can be forwarded or not (operator services calls and emergency services calls are not forwarded).

If the call can be forwarded, the CFV AS MUST re-write the request-URI with the stored forward-to address, insert a history-info header in accordance with Section 7.3, follow the route header in the request and send it back to the S-CSCF.

When a call is forwarded, if the originally targeted UE is subscribed to the ua-profile Event Package for Call Forwarding, the CFV AS MUST send a NOTIFY informing the originally targeted UE that a call has just been forwarded. Upon receipt of the NOTIFY, if the NOTIFY indicates that a new call has been forwarded, the UE MUST play ringsplash if provisioned to provide ring reminder. The UE MUST record the number of calls forwarded since CFV was activated, for comparison to the value in potential future NOTIFY messages.

If the CFV AS determines that the call cannot be forwarded, the CFV AS MUST return an error response to the requesting UE.

The CFV AS MUST NOT forward the INVITE if the number of sequential forwards allowed is exceeded and MUST return the error response to the requesting UE.

7.3.2.4 Feature Data

Table 21 summarizes the feature data items that are defined in support of CFV.

Table 21 - Call Forwarding Variable Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Forward-to address	SIP or Tel URI	Non-volatile	Per public user ID	AS	AS	AS	None
Feature availability status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
User provided forward-to address	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
Feature activation status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
Verification/courtesy call required indicator	Boolean	Non-volatile	Per network	AS	AS	AS	None
Verification/courtesy call 2nd attempt or answer required indicator (see Note 1)	Boolean	Non-volatile	Per network	AS	AS	AS	None
Verification 1st number attempted (see Note 1)	SIP or Tel URI	Transient	Per public user ID	AS	AS	AS	None
Number of sequential forwards allowed	Integer, 0=disabled	Non-volatile	Per network	AS	AS	AS	None
Forwarding to toll number allowed indicator	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
Forwarding to IDDD allowed indicator	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Forwarding with carrier access code allowed indicator (see Note 2)	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
Special conditions dial-tone when forwarded indicator	Boolean	Volatile on UE, Non-volatile on XDS	Per network	XDS	XDS	UE	Mandatory
Call Forward Ring Reminder	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
Call Forwarding Subscription Duration	Integer (seconds)	Volatile on UE, Non-volatile on XDS	Per public user ID	XDS	XDS	UE	Mandatory
Number of Calls Forwarded Since Activation	Integer	Volatile on UE, Non-volatile on AS	Per public user ID	AS	AS	AS, UE	None
Note 1: Applicable only if verification/courtesy calls are enabled. Note 2: See [PKT 24.229] for rules for populating the Request URI for address digits containing carrier access code.							

Table 22 - Call Forwarding Variable UA-Profile Event Package Notify Body

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Call Forwarding Variable Activation Status	Boolean (0=inactive, 1=active)	Volatile on UE, Non-volatile at AS	Per public user ID	AS	AS	AS, UE	None
Number of Calls Forwarded Since Activation	Integer	Volatile on UE, Non-volatile at AS	Per public user ID	AS	AS	AS, UE	None

7.3.2.5 Feature Interactions

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

Call Blocking

The CFV AS MUST NOT allow the activation of the call forwarding variable to blocked numbers. The CFV AS MUST give error notification in response to this activation request.

Call Forwarding Busy Line (CFBL)

When CFV is deactivated, CFBL takes precedence.

When activated, CFV takes precedence over CFBL.

Call Waiting (CW)

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

Cancel Call Waiting (CCW)

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

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

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

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

IDDD Via an Operator System

The CFV AS MUST NOT allow 010 and 01+ as "forward to" numbers.

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

Emergency Service (911)

The CFV AS SHOULD NOT allow 911 as a "forward to" number.

IC/INC Interconnection

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

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

The CFV AS MUST NOT allow 950-WXXX as a "forward to" number.

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

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

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

Do Not Disturb

Do Not Disturb takes precedence over Call Forwarding Variable.

Solicitor Blocking

Solicitor Blocking takes precedence over Call Forwarding Variable.

Operator Services

The CFV AS MUST NOT allow 0-, 0+, or X11 as a "forward to" number.

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

The CFV AS SHOULD return a 486 to a Busy Line Verify INVITE with Join header.

Selective Call Forwarding

Selective Call Forwarding takes precedence over Call Forwarding Variable

7.3.2.6 Call Flows

Call flows contained in this section illustrate the Call Forwarding Variable (CFV) feature. The call flows are for example purposes only, and are not normative, so if there is a discrepancy between the written text requirements and these call flows, the written text requirements take precedence.

Figure 25 illustrates CFV activation procedure, where the user at the "forward to" address answers the call.

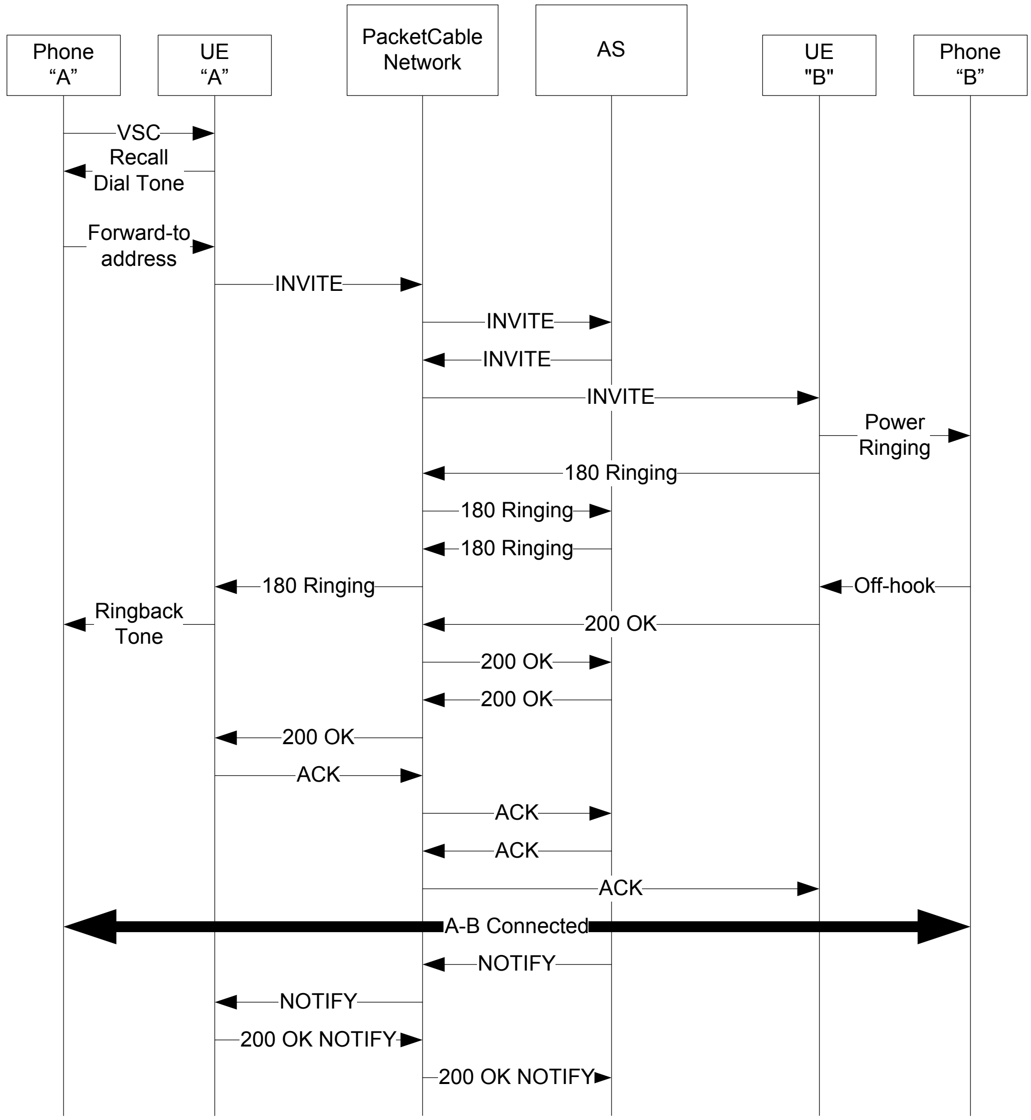

Figure 25 - Call Forwarding Variable (CFV); Activation with User-provided Address

Figure 26 illustrates CFV activation procedure, where the user at the "forward to" address does not answer the call.

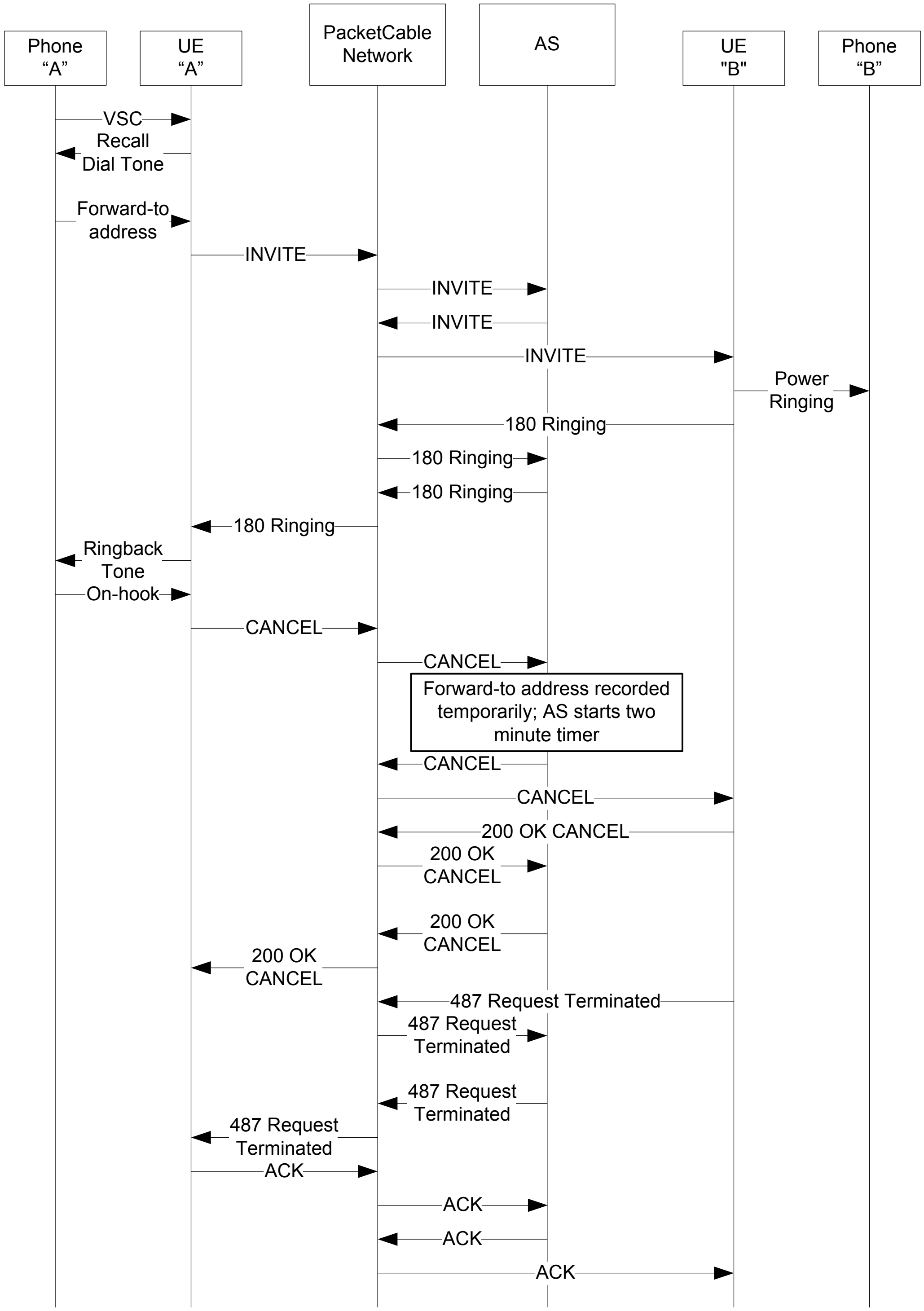

Figure 26 - CFV; Activation with User-Provided Address; No Answer at Fwd-to Address

Figure 27 illustrates the CFV activation procedure, in which the user at the "forward to" address does not answer the call, and the subscriber makes a second attempt to the same number within two minutes of the first attempt.

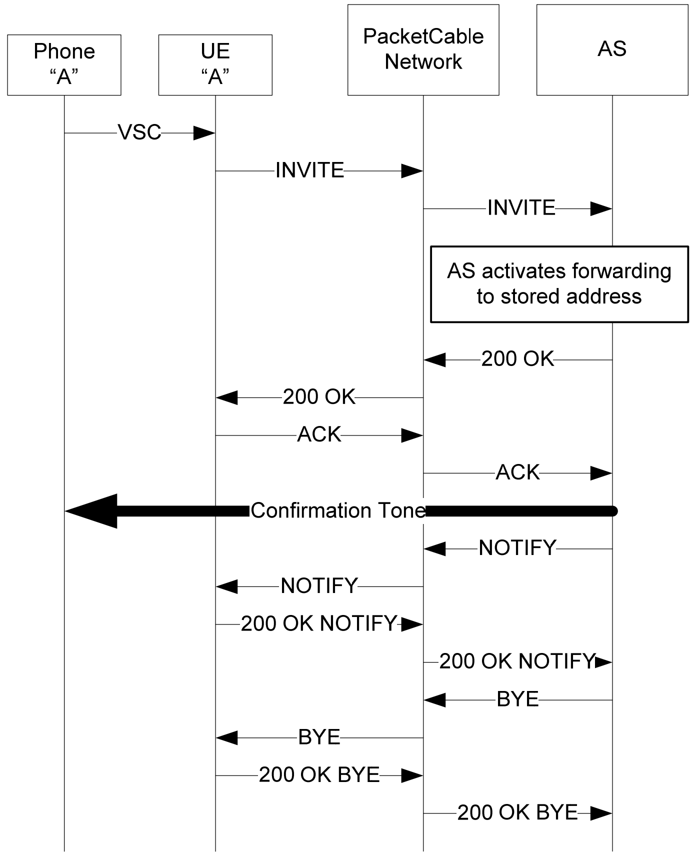

Figure 27 - CFV; Activation with User-provided Address; Second Attempt within Two Minutes to the Same Address

Figure 28 illustrates CFV activation procedure, where verification call is not enabled.

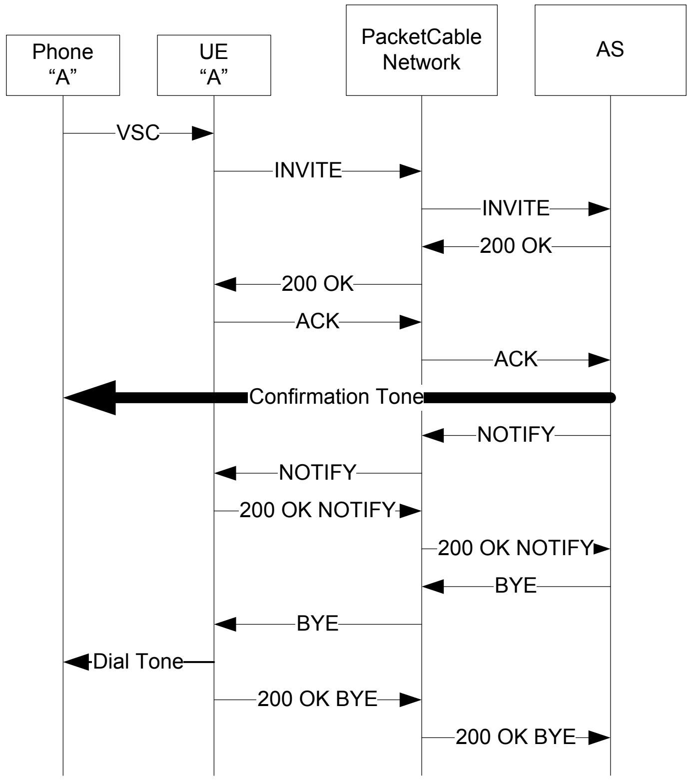

Figure 28 - CFV; Activation to a Fixed Number

Figure 29 illustrates CFV deactivation procedure.

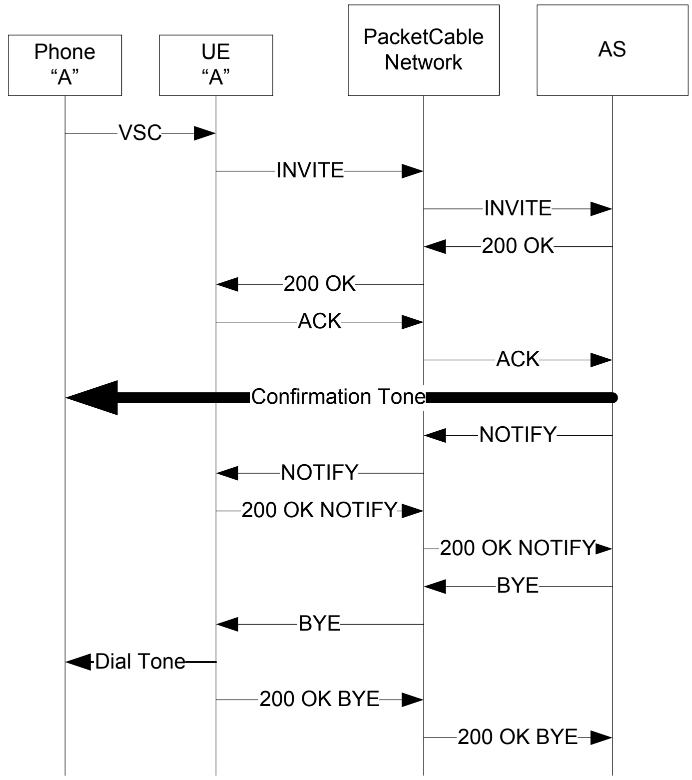

Figure 29 - CFV; Deactivation

Figure 30 illustrates a call being forwarded when the CFV feature is active for the callee.

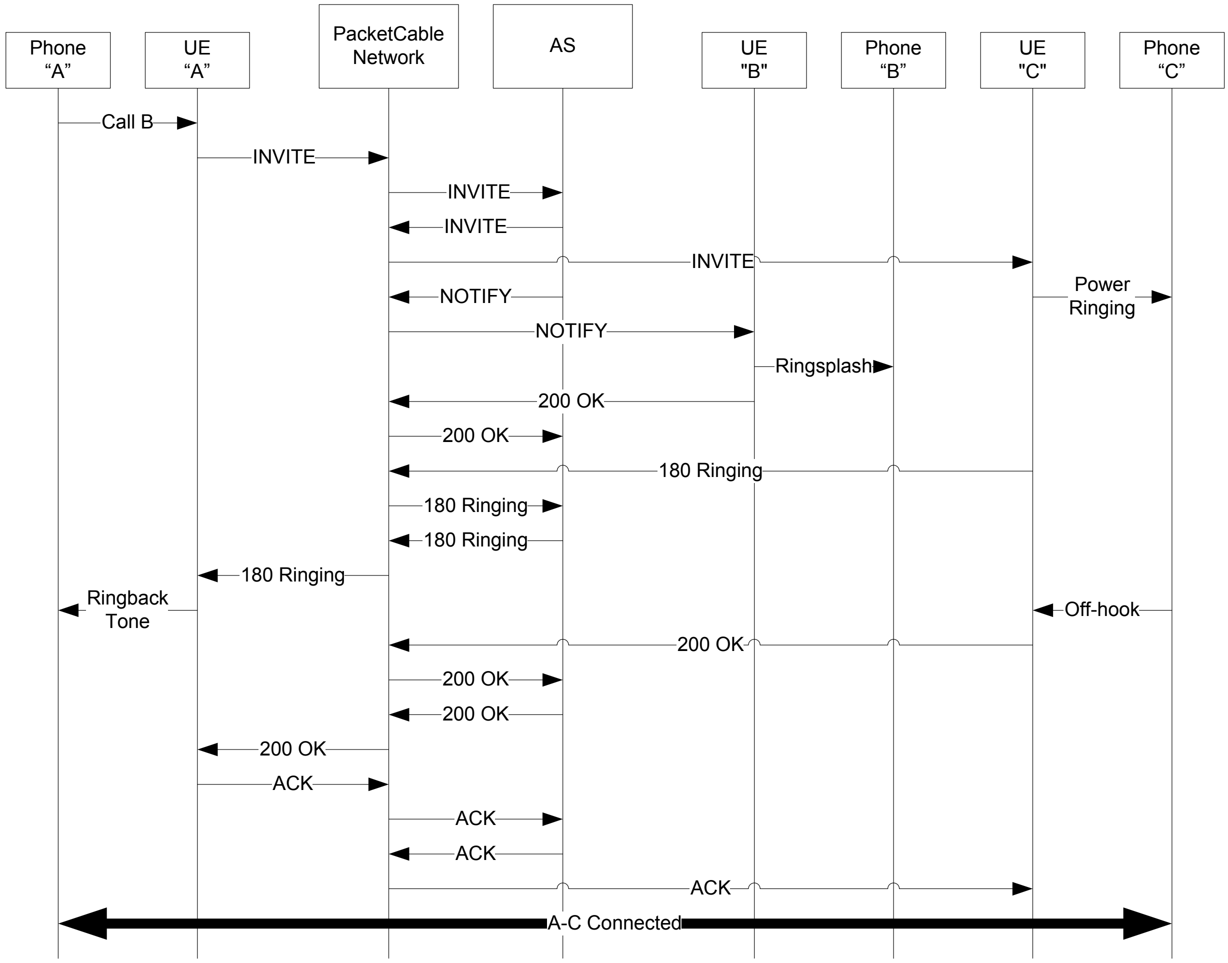

Figure 30 - CFV; Forwarded Call

7.3.3 Call Forwarding Don't Answer

7.3.3.1 Feature Description

Call Forwarding Don't Answer (CFDA) is a feature that allows forwarding of all unanswered calls to the subscriber's public identity to another location. With CFDA, the forwarding happens after a pre-provisioned timeout if the forwarded public identity has not answered the call.

The behavior, as described in Telcordia [GR 586], MUST be followed by UEs with analog interfaces, with the following exceptions. The following sections and requirements within normative reference [GR 586] do not apply to PacketCable networks. These deviations are additional to any divergence from [GR 586] detailed elsewhere in this document:

- UEs are not required to support single key activation/deactivation, as described in [GR 586].
- Forwarding restrictions based on source or destination addresses [GR 586] do not apply.
- Tones may be substituted for the announcements described in [GR 586].
- Restriction for forwarding to "free numbers" found in [GR 586] does not apply.

- Charging requirements in [GR 586] do not apply.
- [GR 586] requirements related to business groups do not apply.
- Statements in [GR 586] regarding announcement of forwarding do not apply.
- The comments about gain in [GR 586] do not apply.
- Traffic and maintenance measurements requirements in [GR 586] do not apply.
- The maintenance requirements in [GR 586] do not apply.
- Statements in [GR 586] regarding no-answer on local switch do not apply.

UEs with non-analog interfaces are allowed to provide alternative user interfaces because different types of UEs have widely varying user interface capabilities. In addition to the wide range of UEs, an operator may offer a web client mechanism to allow a subscriber to manage his or her service data. For this section, however, descriptions are limited to interactions with the analog interface telephony client, rather than a service management web client.

7.3.3.2 Feature Activation and Deactivation

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

7.3.3.3 Feature Execution

For calls placed to a public identity that is subscribed to CFDA, the INVITE is proxied to a CFDA AS. Upon receipt of the initial INVITE, the CFDA AS MUST forward the INVITE to the target UE. Upon receipt of a 180 Ringing response to the INVITE, the CFDA AS MUST start the no answer timer.

If the no answer timer expires before a 200 OK to the INVITE is received, the CFDA AS MUST send a CANCEL to the target UE, and determine whether this call can be forwarded (for example, operator services calls and emergency services calls are not forwarded). If the call can be forwarded, the CFDA AS MUST re-write the request-URI with the provisioned forward-to address, insert a history-info header in accordance with Section 7.3, follow the route header in the request, and send the INVITE back to the S-CSCF.

If the CFDA AS fails to receive any response to the INVITE from the target UE within the SIP response timeout, the CFDA AS MUST determine whether this call can be forwarded (for example, operator services calls and emergency services calls are not forwarded). If the call can be forwarded, the CFDA AS MUST re-write the request-URI with the provisioned forward-to address, insert a history-info header in accordance with Section 7.3, follow the route header in the request, and send the INVITE back to the S-CSCF.

If the CFDA AS determines that the call cannot be forwarded, the CFDA AS MUST return an error response to the requesting UE.

The CFDA AS MUST NOT forward the INVITE if the number of sequential forwards allowed is exceeded and MUST return the error response to the requesting UE.

7.3.3.4 Feature Data

Table 23 summarizes the feature data items that are defined in support of CFDA.

Table 23 - Call Forwarding Don't Answer Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Forward-to address	SIP or Tel URI	Non-volatile	Per public user ID	AS	AS	AS	None
Feature availability status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
No answer timeout	Integer (seconds)	Non-volatile	Per public user ID	AS	AS	AS	None
Number of sequential forwards allowed	Integer, 0=disabled	Non-volatile	Per network	AS	AS	AS	None

7.3.3.5 Feature Interactions

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

Call Forward Variable (CFV)

CFV takes precedence over CFDA.

Call Forward Busy Line (CFBL)

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

In the instance where multiple UEs exist behind the public identity, in which at least one is busy and at least one is idle, the CFDA AS MUST determine this to be a no answer scenario, not a busy scenario.

Call Waiting (CW)

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

Call Blocking

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

Do Not Disturb

Do Not Disturb takes precedence over CFDA.

Selective Call Forwarding

Selective Call Forwarding takes precedence over CFDA.

7.3.3.6 Call Flows

The following call flows illustrate the CFDA feature. The call flows are not normative, so if there is a discrepancy between the written text requirements and these call flows, the written text requirements take precedence.

Figure 31 illustrates a call being forwarded due to No Answer timeout.

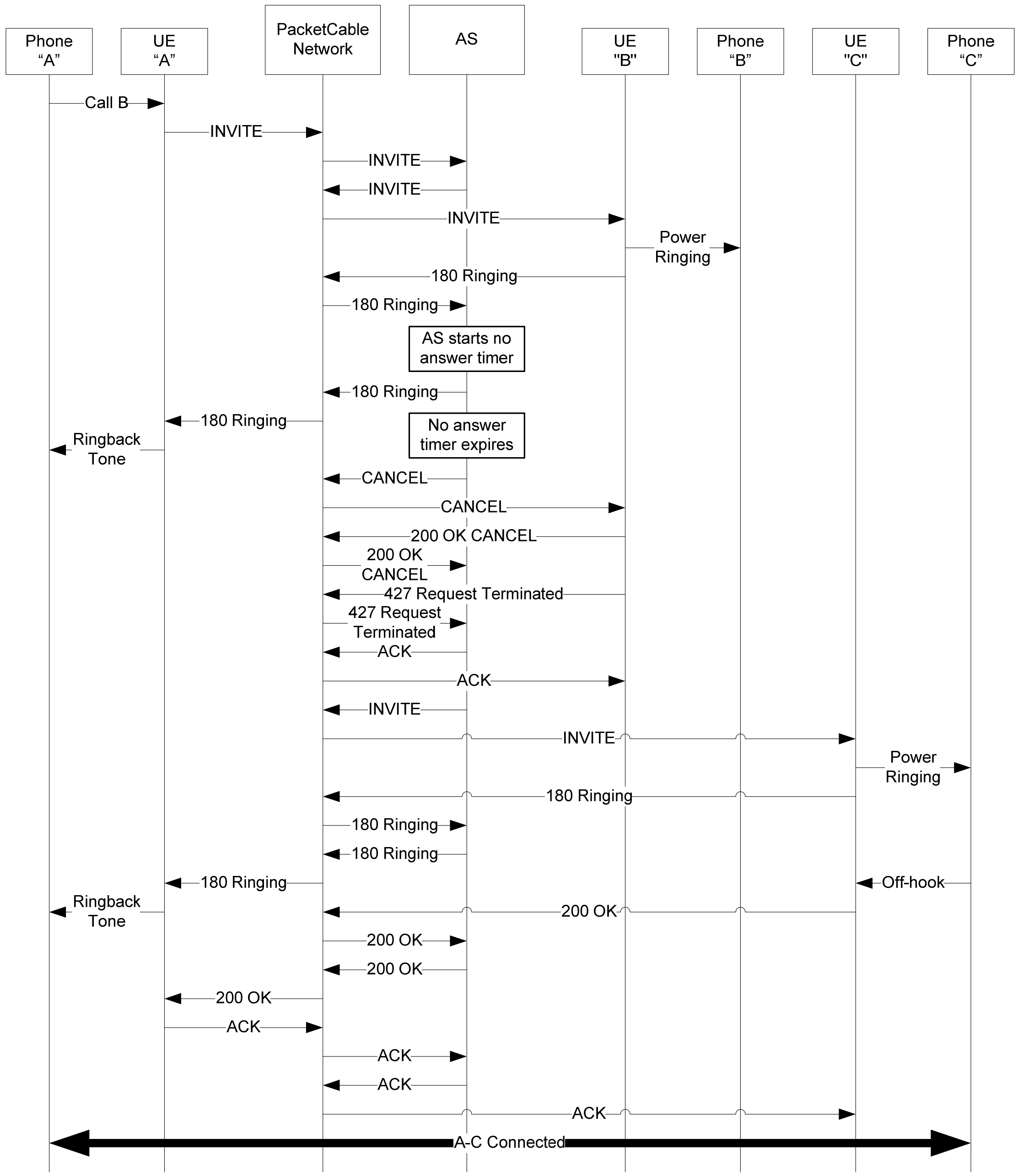

Figure 31 - Call Forward Do Not Answer (CFDA); Establishment Due to No Answer Timeout

Figure 32 illustrates a call being forwarded due to SIP response timeout.

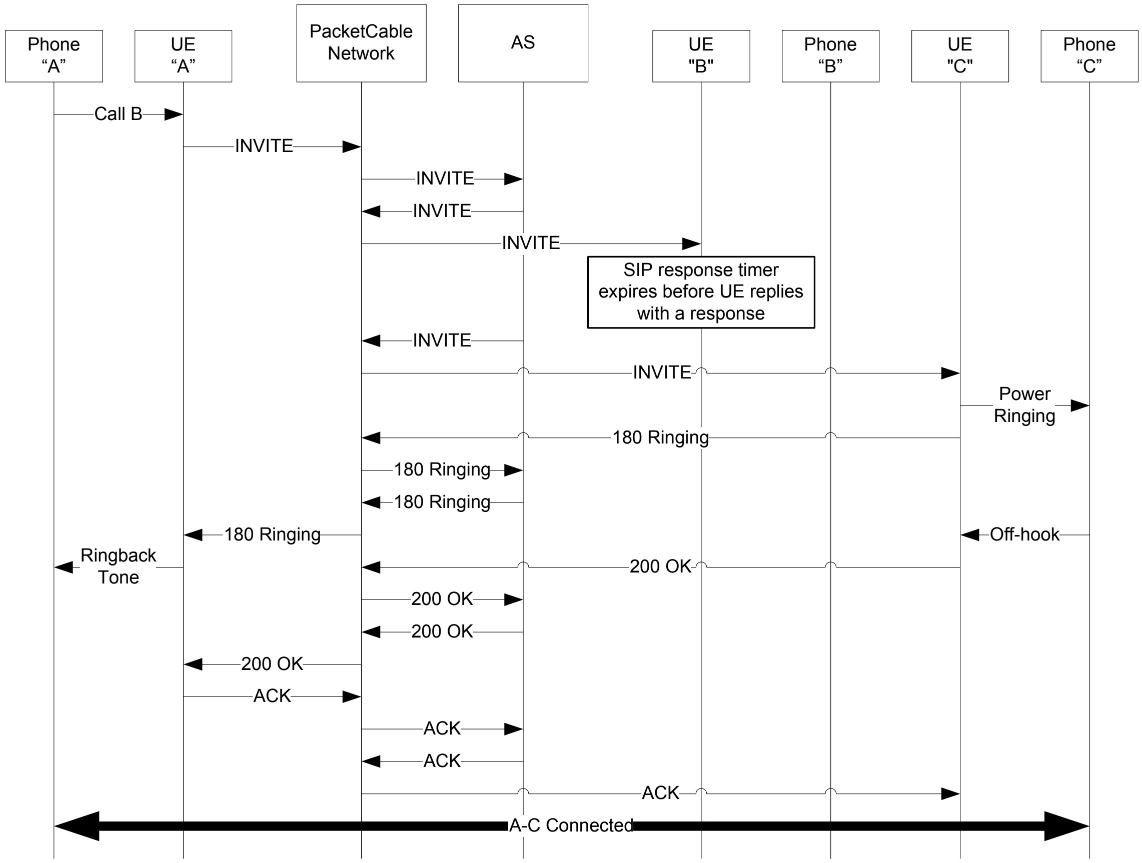

Figure 32 - CFDA; Establishment Due to SIP Response Timeout

7.3.4 Call Forwarding on Busy

7.3.4.1 Feature Description

Call Forwarding Busy Line (CFBL) is a feature that allows a subscriber to forward calls addressed to the subscriber's public identity to another location when the subscriber is not capable of receiving another incoming call. For CFBL, the forward-to address is configured by the network operator, so, unlike CFV, the subscriber cannot designate the forward-to address by dialing a feature activation code and entering a forward-to address. With CFBL the forwarding happens immediately upon detection that the public identity cannot receive the call and the forwarded public identity has no opportunity to answer the call.

The behaviors, as described in Telcordia [GR 586], are followed by UEs with analog interfaces, with the following exceptions. The following sections and requirements within normative reference [GR 586] do not apply to PacketCable networks. These deviations are additional to any divergence from [GR 586] detailed elsewhere in this document:

- UEs are not required to support single key activation/deactivation, as described in [GR 586].
- Forwarding restrictions based on source or destination addresses, as described in [GR 586], do not apply.
- Tones may be substituted for the announcements, described in [GR 586].
- Restriction for forwarding to "free numbers" found in [GR 586] does not apply.

- Charging requirements in [GR 586] do not apply.
- [GR 586] requirements related to business groups do not apply.
- Statements in [GR 586] regarding announcement of forwarding do not apply.
- The comments about gain in [GR 586] do not apply.
- Traffic and maintenance measurements requirements in [GR 586] do not apply.
- The maintenance requirements in [GR 586] do not apply.
- Statements in [GR 586] regarding no-answer on local switch do not apply.

7.3.4.2 Feature Activation and Deactivation

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

7.3.4.3 Feature Execution

For calls placed to a public identity that is subscribed to CFBL, the INVITE is proxied to a CFBL AS. Upon receipt of the initial INVITE, the CFBL AS MUST forward the INVITE to the target UE. If the CFBL AS receives a 486 BUSY in response to the INVITE, the CFBL AS determines whether the call can be forwarded by checking the feature availability status for the target UE and for the presence of a forward-to-address. If the CFBL AS determines the call is eligible for forwarding, the CFBL AS MUST replace the request URI in the original INVITE with the forward-to address, and forward the INVITE to the new target.

If the CFBL AS determines that the call cannot be forwarded, the CFBL AS MUST return the error response to the requesting UE.

If the CFBL AS does not receive a response to the forwarded INVITE, the CFBL AS MUST NOT invoke the CFBL procedures and interprets this as a Call Forward Don't Answer (CFDA) condition.

For all other responses to the forward INVITE, the CFBL AS MUST proxy the response back to the requesting UE.

The CFBL AS MUST NOT forward the INVITE if the number of sequential forwards allowed is exceeded and MUST return the error response to the requesting UE.

7.3.4.4 Feature Data

Table 24 summarizes the feature data items that are defined for the implementation of CFBL.

Table 24 - Call Forwarding Busy Line Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Forward-to address	SIP or Tel URI	Non-volatile	Per public user ID	AS	AS	AS	None
Feature availability status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
Number of sequential forwards allowed	Integer, 0=disabled	Non-volatile	Per network	AS	AS	AS	None

7.3.4.5 Feature Interactions

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

Call Forward Variable (CFV)

CFV takes precedence over CFBL.

Call Waiting (CW)

If a CFBL subscriber is also subscribed to CW, then the UE MUST invoke the Call Waiting procedures, as defined in Section 7.5.2, for the first additional call offered. Further, if the UE is handling two calls under CW, the UE MUST return a 486 Busy response to subsequent INVITE requests.

Call Blocking

The remote address is subject to outbound call blocking restrictions.

Operator Services

Busy Line Verification takes precedence over CFBL.

Emergency Interrupt takes precedence over CFBL.

Do Not Disturb (DND)

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

7.3.4.6 Call Flows

The following call flow illustrates the CFBL feature. The call flows are not normative, so if there is a discrepancy between the written text requirements and these call flows, the written text requirements take precedence.

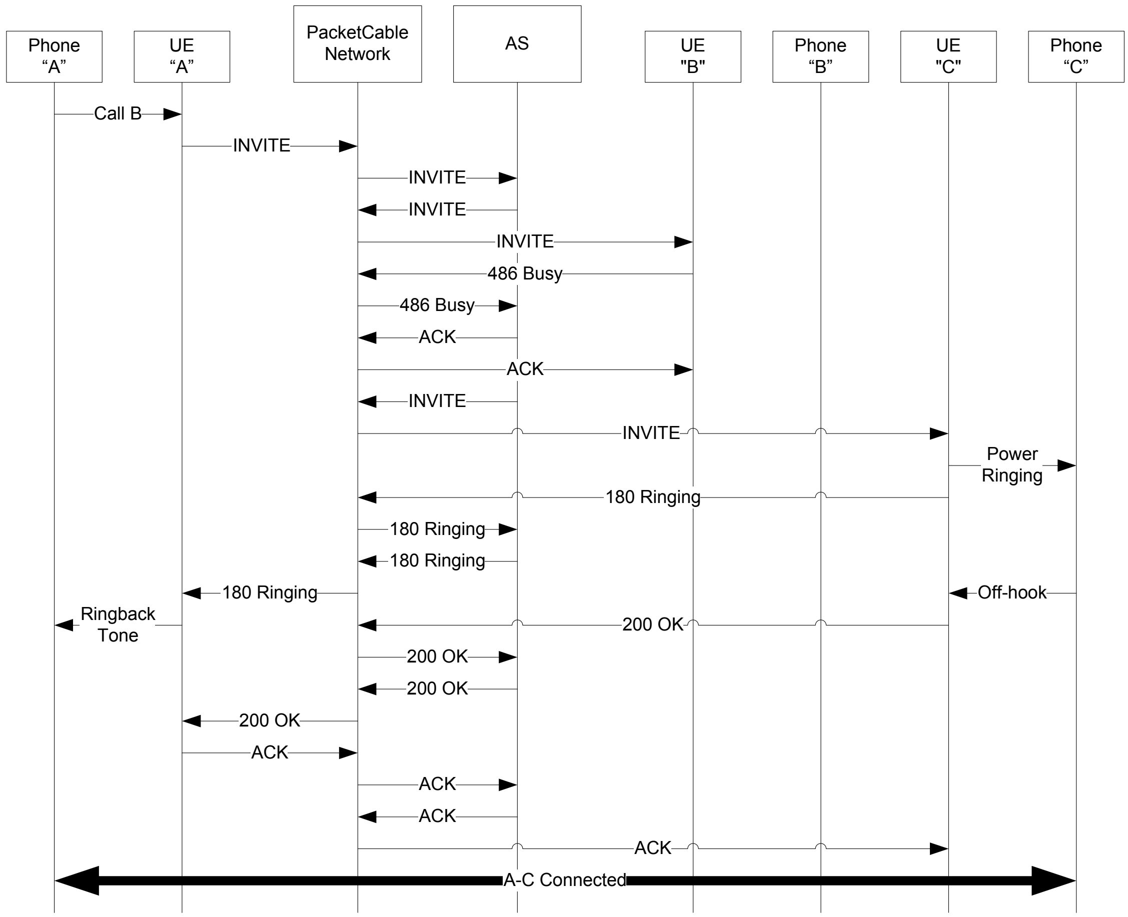

Figure 33 - Call Forward Busy Line (CFBL); Establishment

7.3.5 Selective Call Forwarding

7.3.5.1 Feature Description

Selective Call Forwarding (SCF) is an incoming call management feature that allows customers to define a special list of addresses and remote addresses. Incoming calls from public identities that are on the list are forwarded to the remote address. With SCF, the forwarding happens immediately and the original target public identity has no opportunity to answer the call.

SCF differs from the other call forwarding features in that it is dependent upon the Screening List Editing (SLE) feature. A subscriber activates, deactivates, and configures the SCF feature by dialing a vertical service code (VSC) (typically *63 and/or *83). Each VSC typically provides the subscriber with access to the same set of SCF capabilities. This allows service providers that currently have subscribers using *63 and *83 as separate activation and deactivation codes to continue such use while permitting other service providers that are not currently providing screening list features to advertise just one VSC.

The behavior, as described in Telcordia [GR 217], are met by UEs with analog interfaces. The following sections and requirements within normative reference [GR 217] do not apply to PacketCable networks. These deviations are additional to any divergences from [GR 217] detailed elsewhere in this document:

- Forwarding restrictions to "free numbers" do not apply.
- Charging requirements do not apply.

- The billing requirements in [GR 217] do not apply.
- [GR 217] requirements related to business groups do not apply.
- Traffic and maintenance measurements requirements do not apply.
- The maintenance requirements do not apply.

UEs with non-analog interfaces are allowed to provide alternative user interfaces because different types of UEs have widely varying user interface capabilities. In addition to the wide range of UEs, an operator may offer a web client mechanism to allow a subscriber to manage his or her service data. For this section, however, descriptions are limited to interactions with the analog interface telephony clients, rather than service management web clients.

7.3.5.2 Feature Activation and Deactivation

The user can activate or deactivate the DND feature via the telephony user interface (that is, the phone keypad) by entering a VSC. The SCF VSCs are defined for the UE as part of the Digit Map defined in Section 7.1.1. Upon matching the user-entered digits with the SCF VSCs, the UE carries out the SCF-PROGRAM action. If the feature availability status indicates that the SCF feature is not available, then the UE MUST NOT send the VSC to the SCF AS. If the feature availability status indicates that the SCF feature is available, then the UE MUST send the VSC to the SCF AS via the mechanism specified in Section 7.1.2. The SCF AS MAY, in turn, connect the user to an IVR system for SCF programming and activation and deactivation.

As part of the SCF programming operation, the SCF AS MUST provide the user with an SLE feature, as described in Section 7.6.7, to program a list of callers (that is, their public user IDs) who are subject to SCF treatment when the SCF feature is in effect. The list of callers is stored in the Screen List attribute defined in Table 25. Further, the user has the ability to disable the use of SCF Screen List through the use of the Feature Activation Status attribute defined in Table 25.

At boot time, if Feature Availability Status attribute indicates the SCF feature is available to the UE, the UE MUST SUBSCRIBE to the ua-profile Framework event package on the SCF AS.

When the activation state of SCF feature changes, the SCF AS MUST send a NOTIFY to the UE indicating the activation state of SCF as defined in Table 25.

Once the remote address is specified, the SCF AS MUST store the forward-to address in such a manner that it remains persistent even in the case of power loss in the UE or the SCF AS. Similarly, the SCF AS MUST store the screening list for SCF in such a manner that it remains persistent even in the case of power loss in the UE or SCF AS.

7.3.5.3 Feature Execution

Upon receipt of an INVITE, the SCF AS decides if the INVITE needs to be presented to the called UE based on the current status of the SCF feature and whether the calling party is present in the Screen List. The current feature status is determined by the feature provisioning information and the activation/deactivation status previously set by the user (via the telephony user interface or the web portal).

If the SCF feature is not active, the SCF AS MUST forward the INVITE to the called party.

If the SCF feature is active, the SCF AS tries to match the calling party's public user ID with an entry in the Screen List. If a match is established, the SCF AS MUST forward the INVITE to the forward-to address by rewriting the request URI of the INVITE to that of the forward-to address, insert a history-info header in accordance with Section 7.3, and preserve the originating calling line identification. An address can exist on an SCF list as a Tel URI in an extension format (when applicable), in its 7/10 digit format, in both formats (that is, two entries on the list) or as a SIP URI. When the address of an incoming call is compared to the customer's SCF list, a match should be found if that address exists on the list in any format, regardless of the format of the incoming address.

Further, if the UE is subscribed to the ua-profile Event Package for call forwarding, then the SCF AS MUST send a NOTIFY request informing the originally targeted UE that a call has just been forwarded. The SCF AS MUST set the contents of the NOTIFY according to the data defined in Table 26. If the NOTIFY received by the UE indicates that a new call has been forwarded, the UE MUST play ringsplash if the Call Forward Ring Reminder is set.

If the SCF feature is active and a match is not found, the SCF AS MUST forward the INVITE to the called party.

7.3.5.4 Feature Data

Table 25 summarizes the feature data items that are defined for the implementation of SCF.

Table 25 - SCF Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Forward-to address	SIP or Tel URI	Non-volatile	Per public user ID	AS	AS	AS	None
Feature availability status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
Feature activation status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
Screening List	List of SIP or Tel URIs	Non-volatile	Per public user ID	AS	AS	AS	None
Number of sequential forwards allowed	Integer, 0=disabled	Non-volatile	Per network	AS	AS	AS	None
Forwarding to toll number allowed indicator	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
Forwarding to IDDD allowed indicator	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
Forwarding with carrier access code allowed indicator	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
Call Forward Ring Reminder	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None

Table 26 - Selective Call Forwarding UA-Profile Event Package Notify Body

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Selective Call Forwarding Activation Status	Boolean (0=inactive, 1=active)	Non-volatile	Per public user ID	AS	AS	AS	None
Number of Calls Forwarded Since Activation	Integer	Non-volatile	Per public user ID	AS	AS	AS	None

7.3.5.5 Feature Interactions

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

Call Blocking

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

Speed Dialing

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

IC/INC Interconnection

The SCF AS MUST NOT allow forward-to addresses to be 950-WXXX numbers.

When a call is forwarded using an Interexchange or International Carrier (IC/INC), the SCF AS MUST provide the address of the forwarding UE to the carrier for billing purposes.

Call Forwarding

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

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

Distinctive Ringing/Call Waiting (DRCW)

SCF takes precedence over Distinctive Ringing/Call Waiting.

Do Not Disturb (DND)

SCF takes precedence over Do Not Disturb.

Call Waiting (CW)

SCF takes precedence over Call Waiting.

Automatic Callback (AC)

If a person attempts to activate AC to a UE that has SCF active, the AC customer's address is checked against the SCF list. If the address is on the list, SCF AS MUST reject the AC request since the status of the remote line cannot be checked. If the number does not match the list, the SCF AS MUST process the INVITE as described in Section 7.3.5.3.

Automatic Recall (AR)

If a person attempts to activate AR to a line that has SCF active, the AR customer's address is checked against the SCF list. If the address is on the list, the SCF AS MUST reject the AC request since the status of the remote line cannot be checked. If the number does not match the list, the SCF AS MUST process the INVITE as described in Section 7.3.5.3.

7.3.5.6 Call Flows

Figure 34 illustrates selective call forwarding. The call flows are not normative, so if there is a discrepancy between the written text requirements and these call flows, the written text requirements take precedence.

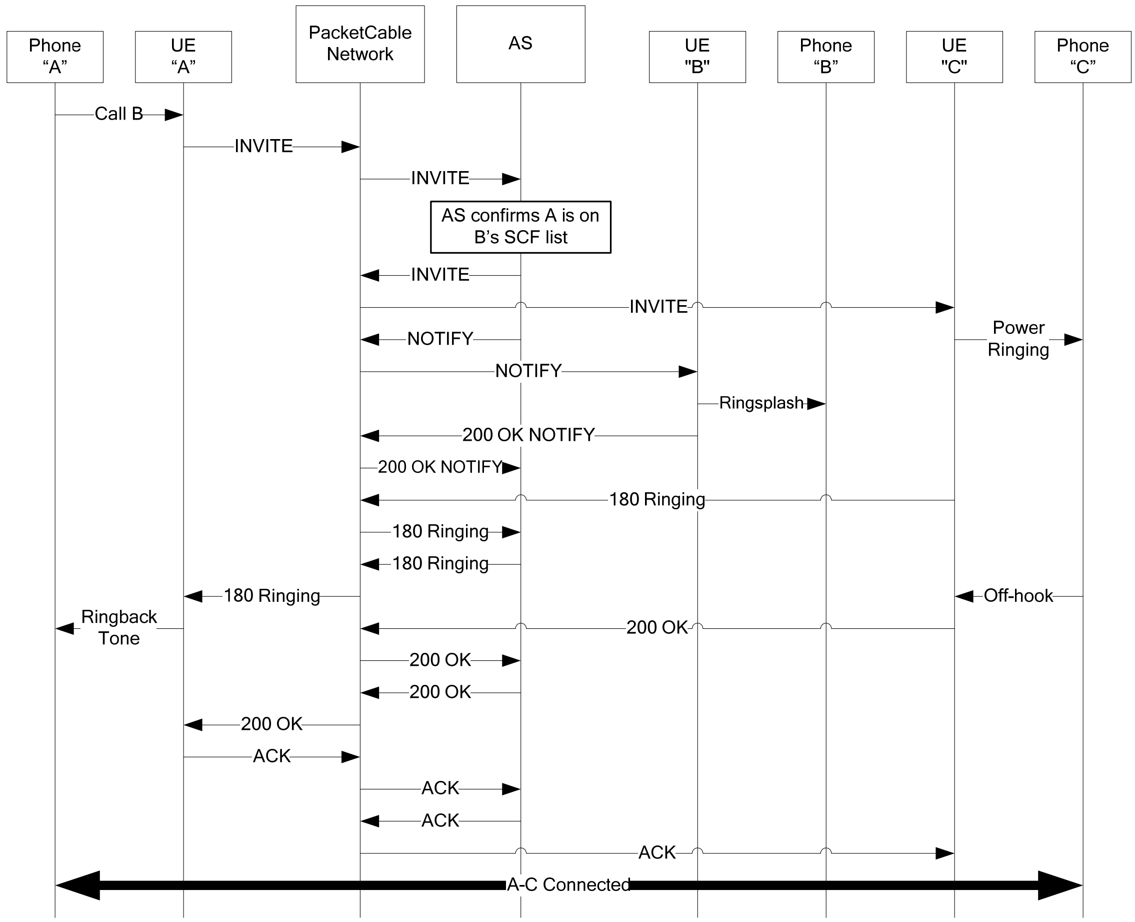

Figure 34 - Selective Call Forwarding (SCF) - Forwarded Call

7.3.6 Remote Activation of Call Forwarding

7.3.6.1 Feature Description

The Remote Activation of Call Forwarding (RACF) feature allows a subscriber who also subscribes to call forwarding variable to control CFV from another location (not from 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 PIN numbers using the SPP feature.

UEs with non-analog interfaces are allowed to provide alternative user interfaces because different types of UEs have widely varying user interface capabilities. In addition to the wide range of UEs, an operator may offer a web client mechanism which allows a subscriber to manage his or her service data. For this section, however, descriptions are limited to interactions with the analog interface telephony client, rather than a service management web client.

7.3.6.2 Feature Activation and Deactivation

To manage RACF, the subscriber places a call to the RACF service access address and follows the RACF service prompts provided by the RACF AS.

The RACF AS activation and deactivation procedures follow those for CFV. Thus, the RACF AS MUST implement the CFV procedures as defined in Section 7.3.2.2.

If RACF programming fails for any reason, the RACF AS MUST NOT change the CFV status.

If a subscriber successfully deactivates CFV via RACF, but CFV was not activated in the first place, the RACF AS MUST play a confirmation indication to the subscriber anyway.

If a subscriber already has CFV activated when he or she successfully programs CFV via RACF, the new forward-to address provided via the RACF programming overrides the pre-existing CFV programming and the UE MUST provide a confirmation indication to the subscriber.

If a subscriber dials an invalid forward-to address while programming CFV via RACF (as described in the CFV section), the RACF AS MUST prompt the subscriber to enter a different address.

If the RACF AS receives a BYE from the UE prior to the confirmation tone on an RACF programming attempt, the RACF AS MUST NOT change the CFV status.

7.3.6.3 Feature Execution

RACF is solely an activation and deactivation feature. There is no call processing impact for RACF.

7.3.6.4 Feature Data

Table 27 summarizes the feature data items that MUST be supported for the implementation of RACF.

Table 27 - Remote Activation of Call Forwarding Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Feature availability status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
RACF PIN	Digits	Non-volatile	Per public user ID	AS	AS	AS	None

7.3.6.5 Feature Interactions

Call Forward Variable

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

The RACF subscriber is also subscribed to CFV.

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

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

7.3.7 Call Forwarding to Voice Mail

Call forwarding to voice mail is achieved by the service provider provisioning the CFDA and CFBL features with a forward-to address of the voice mail system. Some network operators may wish to allow the voice mail system to know why the call was forwarded (busy versus no answer, for example). The CF AS MAY insert reason information in the INVITE response per [RFC 4458].

7.4 Call Blocking Features

7.4.1 Outbound Call Blocking

Outbound Call Blocking (OCB) prevents UEs from making calls to specific public identities. Service providers typically offer this feature as one or more named outbound call blocking services. Examples include International Call Blocking, Local Directory Assistance Call Blocking, Long Distance Directory Assistance Call Blocking, 900/976 Call Blocking, and Toll Call Blocking.

The digit map syntax supports the ability to allow outbound call blocking for the following dial string types:

- 900/976 Blocking (1900 and 1976)
- Direct-dialed international calls (011+)
- Direct-dialed calls (1+)
- Operated-assisted calls (0+)
- Operator-assisted international calls (01+)
- Operator-request calls (0-, that is, 0 followed by a # or 4-second timeout)
- Local directory assistance calls (411)
- Long distance directory-assistance calls (555)
- Carrier access code (+10XXX)
- Any destination that can be represented as a dial string

7.4.1.1 Feature Description

AS-Based OCB requires that the OCB AS receive each INVITE transmitted by a subscribing UE. The AS determines if the destination URI provided in the INVITE is to a blocked public ID. The AS forwards the INVITE toward its intended destination if it is not addressed to a blocked public ID. If addressed to a blocked public ID, the OCB AS establishes an early media session with the calling UE and announces the call was blocked.

The OCB AS also supports a PIN override option, which is activated or deactivated through provisioning. If the PIN override option is activated, the OCB AS includes an override PIN announcement in the early media session. The caller entered override PIN is forwarded to the OCB AS per the method negotiated in the early media session SDP offer-answer exchange. If the override PIN is authenticated, the OCB AS forwards the INVITE to the destination Public Identity. If the PIN override authentication fails, the OCB AS announces the authentication failure and terminates the early media session by sending a Forbidden (403) response.

If the PIN override option is not activated, the OCB AS sends a Forbidden (403) response after announcing the call has been blocked. The Forbidden (403) response and acknowledgement ends the early media session.

7.4.1.2 Feature Activation and Deactivation

An OCB subscriber can activate or deactivate call blocking or call blocking options using either of the following methods:

- User communications with a service provider operator – This method requires subscriber communication with the service provider. The service provider executes the subscriber's requested OCB changes. The processes for implementing provisioning changes can be unique to each service provider.
- Web Portal Access – This is an automated method whereby the subscriber accesses a web portal to activate, deactivate, or modify OCB subscription options. The processes for accessing the portal and implementing the changes can be unique to each service provider.

7.4.1.3 Feature Execution

Upon receiving an INVITE from the subscriber UE, the OCB AS MUST compare the destination URI provided in the INVITE to provisioned digit maps that identify the blocked Public ID destinations for the subscriber. If a destination Public ID does not match any of the provisioned OCB digit maps, the AS MUST forward the INVITE toward its intended destination. If the destination URI matches a provisioned OCB digit map, the OCB AS MUST execute early media as specified in Section 7.1.9**.** The OCB AS MUST play an early media stream announcing that the call to the destination public identity is blocked

If the AS OCB does not support an override PIN service, it MUST end the early media session by transmitting a Forbidden (403) Response. The session is terminated after an ACK is received from the UE.

When the AS OCB supports a PIN override service, the early media announcement MUST prompt the caller to enter an override PIN. The UE MUST forward the caller's PIN entry per the methods established in the early media SDP offer-answer exchange. The AS MUST compare the collected PIN to the provisioned PIN for the subscriber. If a matching PIN is entered, the AS MUST send the INVITE toward the specified destination URI. If an incorrect PIN is entered, the AS MUST play an announcement indicating that the entered PIN was incorrect. The AS MUST re-attempt to collect the PIN until the correct PIN is entered or the provisioned number of attempts has been reached. The AS MUST terminate the session by sending the originating UE a Forbidden (403) response if the provisioned number of attempts has been reached.

7.4.1.4 Feature Data

Table 28 summarizes the feature data items that are defined to support implementation of OCB feature.

Table 28 - Outbound Call Blocking Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Feature Activation Status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
Caller Treatment Indicator	Strings representing treatment options	Non-volatile	Per public user ID	AS	AS	AS	None
Activation Status for OCB Override PIN	Boolean	Non-volatile	Per network	AS	AS	AS	None
OCB Override PIN	Digits	Non-volatile	Per public user ID	AS	AS	AS	None
Retry Attempts	Integer	Non-volatile	Per public user ID	AS	AS	AS	None

7.4.1.5 Feature Interactions

Speed Dialing and Call Forwarding

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

7.4.1.6 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

Figure 35 illustrates an example call flow where the OCB feature is subscribed to but the feature is presently disabled. The call flow assumes the AS is capable of generating all applicable announcements and RTP streams.

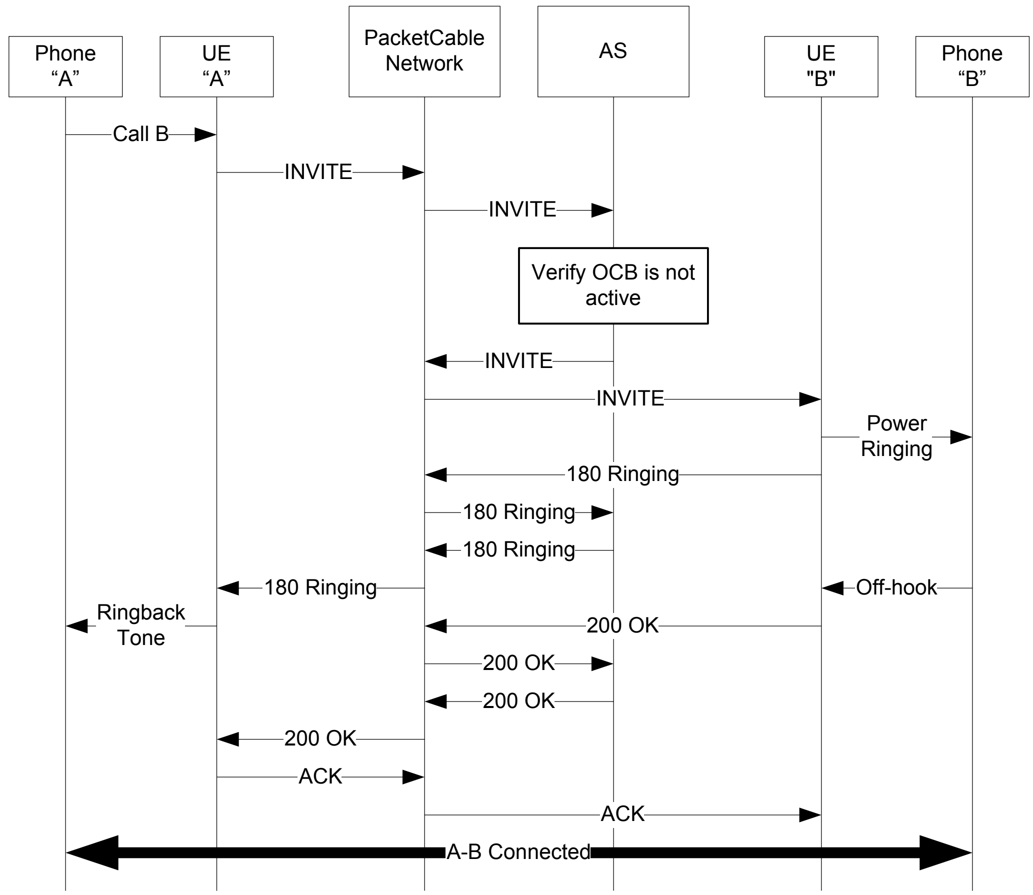

Figure 35 - Outbound Call Blocking (OCB) - Feature Function While Deactivated

Figure 36 illustrates an example call flow where the OCB feature is activated, the destination URI matches a call block category, the originating UE subscribes to an OCB override service, and an invalid override PIN is entered. The call flow assumes the AS is capable of generating all applicable announcements and RTP streams. The call flow is not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

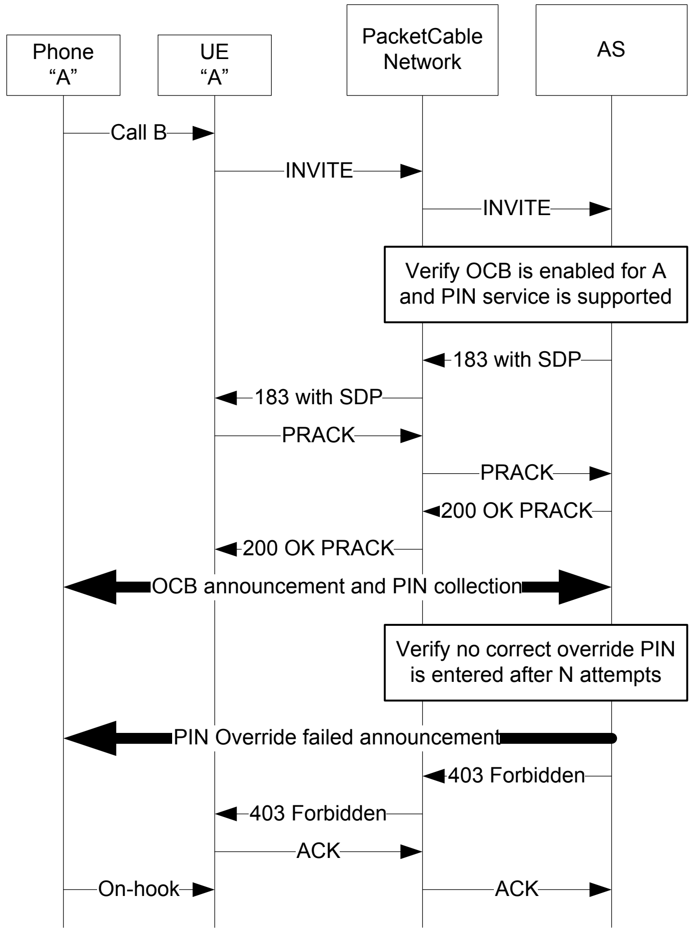

Figure 36 - OCB; Invalid Override PIN Entry

Figure 37 illustrates an example call flow with OCB activated, the destination URI matches a call block category, the originating UE subscribes to an OCB override service, and a valid override PIN is entered. The call flow assumes the AS is capable of generating all applicable announcements and RTP streams. The call flow is not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

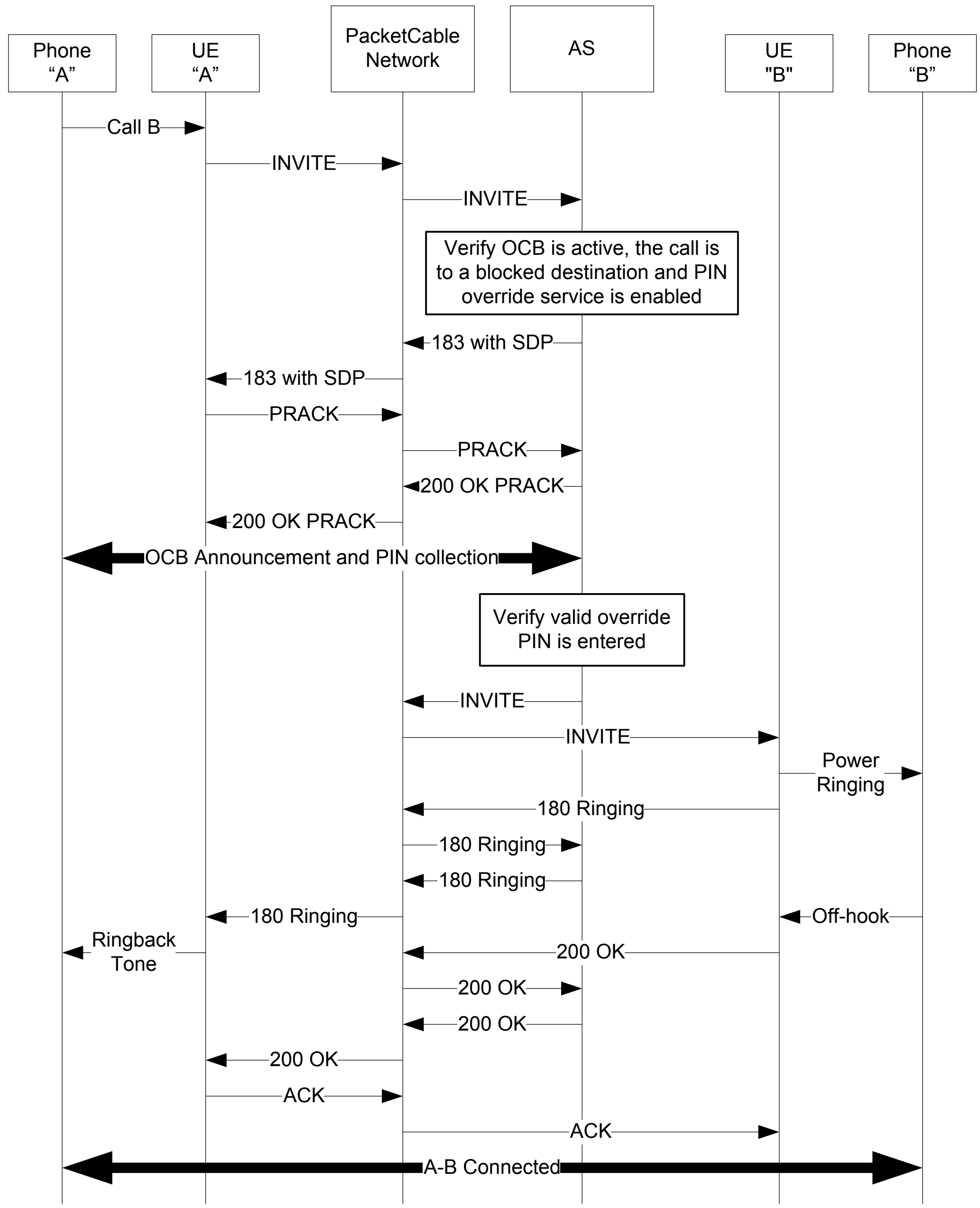

Figure 37 - OCB; Valid Override PIN Entry

Figure 38 illustrates a call flow in which the OCB feature is activated, the destination URI matches a call block category, early media announcement is supported, and there is no override PIN subscription. The call flow assumes the AS is capable of generating all applicable announcements and RTP streams. The call flow is not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

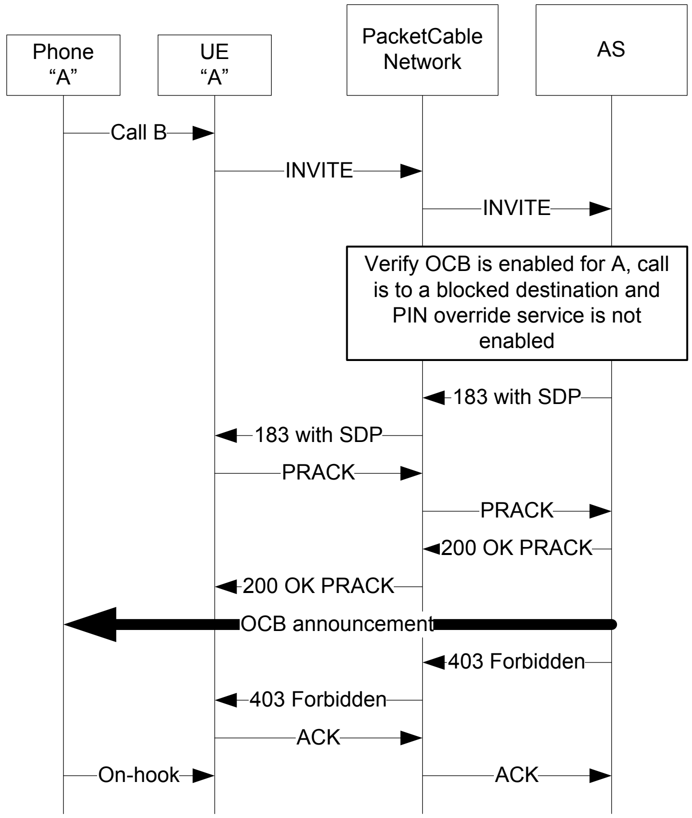

Figure 38 - OCB; Override PIN Service Not Enabled

7.4.2 Collect Call Blocking

Collect Call Blocking prevents termination of a collect call to the subscriber's public identity.

7.4.2.1 Feature Description

A blocked caller receives treatment indicating the subscriber declines to accept the call. Collect Call Blocking is a network based feature relying on LIDB (Line Identification Database) for feature status.

7.4.2.2 Feature Activation and Deactivation

A Collect Call Blocking subscriber can activate or deactivate call blocking or call blocking options using either of the following methods:

- User communications with a service provider operator – This method requires subscriber communication with the service provider. The service provider executes the subscriber's requested Collect Call Blocking changes. The processes for implementing provisioning changes can be unique to each service provider.
- Web Portal Access – This is an automated method whereby the subscriber accesses a web portal to activate, deactivate, or modify Collect Call Blocking subscription options. The processes for accessing the portal and implementing the changes can be unique to each service provider.

7.4.2.3 Feature Execution

The execution of this feature depends on operator interaction with a database external to the PacketCable network. The typical use is as follows:

- A subscriber calls an operator to place a collect call.
- The operator locates the number to call in the LIDB, to determine if the call is possible.
- If the call is possible, the operator places it; otherwise the operator informs the subscriber that the call is blocked.

The call is essentially blocked before it reaches the PacketCable network, requiring no resources from the network.

7.4.2.4 Feature Data

There are no data elements requiring definition in the PacketCable network for this feature.

7.4.2.5 Feature Interactions

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

7.4.3 Solicitor Blocking

Solicitor Blocking (SB) provides an IVR screen between incoming callers and the subscriber. Contrary to its name, this feature can also be used to block callers other than solicitors. Under these circumstances, the feature is generally known as privacy screening.

There are two versions of this feature. In the first version, the incoming caller is connected to an IVR system and played a greeting asking the caller to press a key to be connected to the subscriber, thereby acknowledging the caller is not a solicitor.

In the second version, the SB AS prompts the caller for a name (greeting) to play to the subscriber. The SB AS then calls the subscriber, plays the greeting, and offers a menu of choices for handling the call. The subscriber then selects or rejects the call based on an IVR menu of choices.

In both versions of the feature there is a subscriber specified caller acceptance list of numbers that the subscriber chooses to allow without screening.

Solicitor Blocking is dependent upon the Screening List Editing (SLE) feature to maintain a Solicitor Blocking caller list of numbers that bypasses the screening. A subscriber can initiate procedures for modifying the white list by going off-hook, receiving the dial-tone, and dialing the Solicitor Blocking VSC. Each code provides the subscriber with access to the same set of Solicitor Blocking capabilities.

As part of its SB-MAINT digit map action, if the solicitor blocking feature is available, the UE MUST send an INVITE with the request URI capturing the solicitor blocking VSC as per the network requirements.

Once the solicitor blocking VSC has been successfully entered, the SB AS MUST play announcements providing the following information (not necessarily in this order):

- The name of the service (that is, Solicitor Blocking)
- The current size of the customer's white list
- Actions and associated dialing codes available to the user:
  - Add entr(y)ies to the list
  - Delete entr(y)ies from the list
  - List review

7.4.3.1 Feature Description, Screening by Keypress

A call is placed to a subscriber with Solicitor Blocking. The S-CSCF directs the call to the SB AS. The SB AS begins streaming early media, requesting the caller to enter DTMF digits to indicate the caller is not a solicitor and to connect to the subscriber. If no keypress is detected within a given time the SB AS releases the call with a 480 final response. If the caller presses the accept key, the SB AS transfers the party to the subscriber by initiating a new call. The subscriber's phone rings with the Caller ID info of the caller. The subscriber answers the call and is connected to the caller.

7.4.3.2 Feature Description, Screening by Greeting Record

A call is placed to a subscriber with Solicitor Blocking. The S-CSCF directs the call to the SB AS. The SB AS begins streaming early media, which requests that the calling party leave a greeting to play to the called party. If no message is detected within a given time, the SB AS releases the call with a 480 final response. If the caller records a message, the SB AS initiates a new call to the called party with the intent of playing the collected message. The called party's phone rings with the Caller ID info of the calling party. The called party answers the call and hears the caller's recorded greeting with a selection of options for handling the caller's call (accept, decline, transfer to VM, etc). The calling party is connected to the called party if the called party chooses to accept. The calling party's call is connected to a treatment if the called party chooses to reject.

7.4.3.3 Feature Activation

The Solicitor Blocking feature is always active once assigned to the subscriber.

7.4.3.4 Feature Execution, Screening by Keypress

Upon establishing an early media session with an incoming caller not on the SB caller acceptance list provisioned by the subscriber, the SB AS MUST prompt the caller for keypress and attempt to collect the keypress. The SB AS MUST warn the caller if it fails to detect a keypress, make at least one other attempt to collect, and disconnect the call if the attempt fails. The SB AS MUST allow the call to proceed to the subscriber if the caller is on the SB caller acceptance list. The SB AS MUST forward the caller's INVITE to the subscriber's public identity if the caller presses the accept key.

7.4.3.5 Feature Execution, Screening by Greeting Record

Upon establishing an early media session with an incoming caller not on the SB caller acceptance list provisioned by the subscriber, the SB AS MUST prompt the caller for a greeting and attempt to record the greeting. The SB AS MUST warn the caller if no greeting is detected, make at least one other attempt to get a recorded greeting, and disconnect the call if that attempt fails. The SB AS MUST allow the call to proceed to the subscriber if the caller is on the SB caller acceptance list. The SB AS MUST initiate a new call to the called party if a greeting is recorded. The SB AS MUST play that greeting with an appropriate announcement and list of choices for handling the called party. The SB AS MUST offer accept or decline as options for handling the calling party. The SB AS MAY offer other choices, like a forward to busy greeting or voicemail. Upon acceptance, the SB AS MUST connect the calling and called parties by sending an INVITE with Replaces header to the called party. The Replaces header matches the existing dialog between the SB AS and the called party. The SB AS MUST proxy the final response from the called party to the calling party without record routing. The SB AS MUST terminate the replaced dialog between itself and the called party by sending a BYE to the called party. Upon declination, or upon input timeout, the SB AS MUST play an announcement indicating that the call was declined by the called party (subscriber).

7.4.3.6 Feature Data

Table 29 summarizes the feature data that are defined to support implementation of the Solicitor Blocking feature.

Table 29 - Solicitor Blocking Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Feature availability status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
IVR greeting for collecting caller information	URI for media server	Non-volatile	Per public user ID	AS	AS	AS	None
Acceptance key	Key number	Non-volatile	Per public user ID	AS	AS	AS	None
Solicitor treatment	URI for media server	Non-volatile	Per public user ID	AS	AS	AS	None
Busy treatment	URI for media server	Non-volatile	Per public user ID	AS	AS	AS	None
Voice mail address	SIP URL	Non-volatile	Per public user ID	AS	AS	AS	None
SBC Caller Acceptance List	List of public identities	Non-volatile	Per public user ID	AS	AS	AS	None

7.4.3.7 Feature Interactions

Solicitor Blocking feature interaction is as follows.

Basic Call

Both versions of Solicitor Blocking take precedence over basic calling functionality for call termination.

Incoming calls are rejected with busy treatment while this feature is executing.

Call Forwarding Variable

Both versions of Solicitor Blocking take precedence over Call Forwarding Variable.

Call Forwarding Don't Answer

Both versions of Solicitor Blocking take precedence over CFDA.

CFDA does not execute if the called party fails to answer.

Anonymous Call Rejection

ACR takes precedence over Solicitor Blocking (screening by keypress version) while ACR is active.

Solicitor Blocking (screening by greeting record version) takes precedence over ACR while ACR is active.

Call Waiting

Both versions of Solicitor Blocking take precedence over Call Waiting.

Incoming calls are blocked while the Solicitor Blocking feature executes; therefore, Call Waiting does not execute while the Solicitor Blocking is executing.

Do Not Disturb

Do Not disturb takes precedence over both versions of Solicitor Blocking.

7.4.3.8 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

There are three relevant call flows for Solicitor Blocking: the caller is accepted by the callee, the caller is rejected by the callee, and the caller fails to enter a greeting to play to the callee.

Figure 39 illustrates a call flow in which the caller is accepted by the callee.

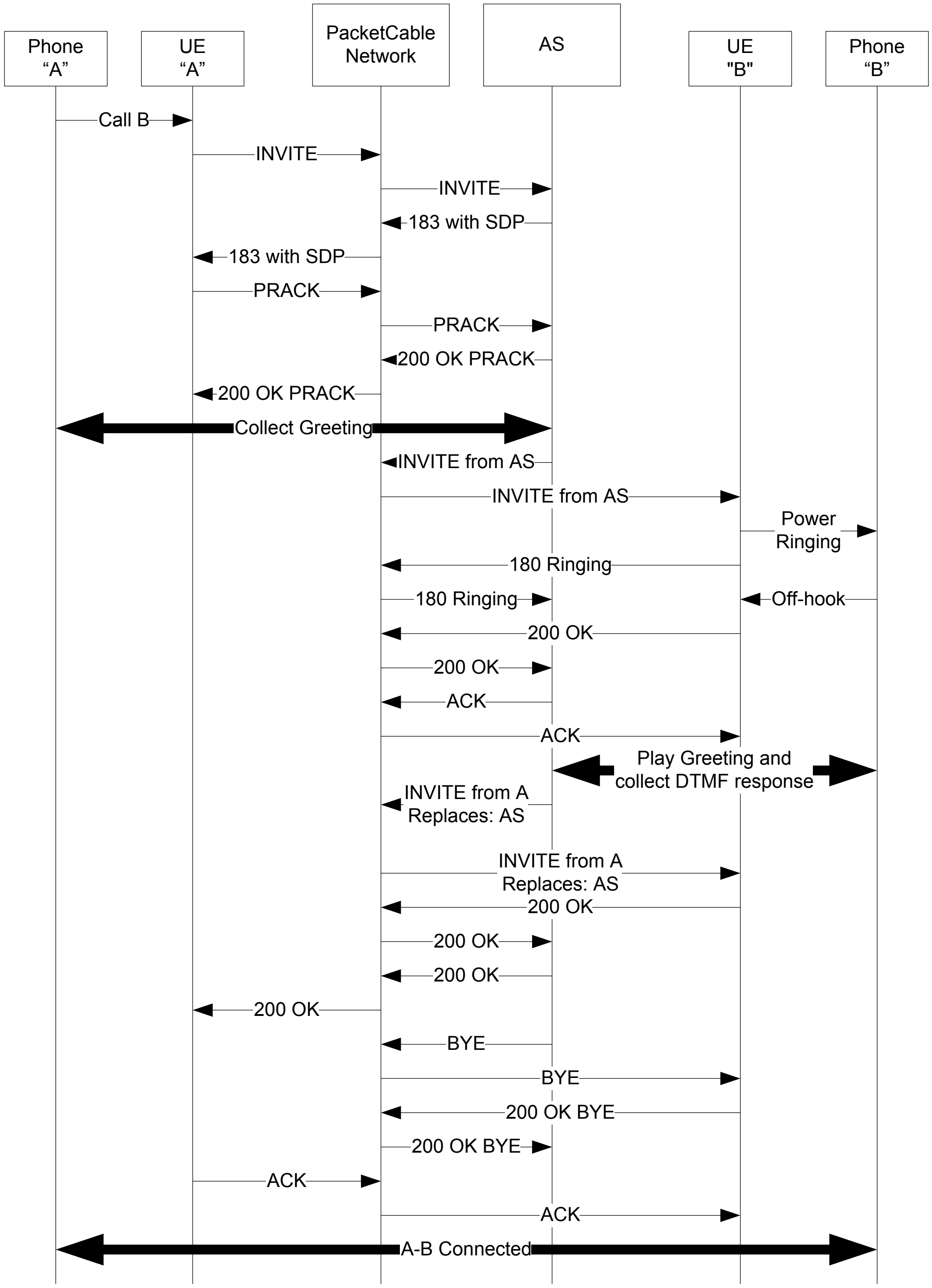

Figure 39 - Solicitor Blocking (SB); Called Party Accepts the Caller

Figure 40 illustrates a call flow in which the callee rejects the caller or fails to respond to the greeting.

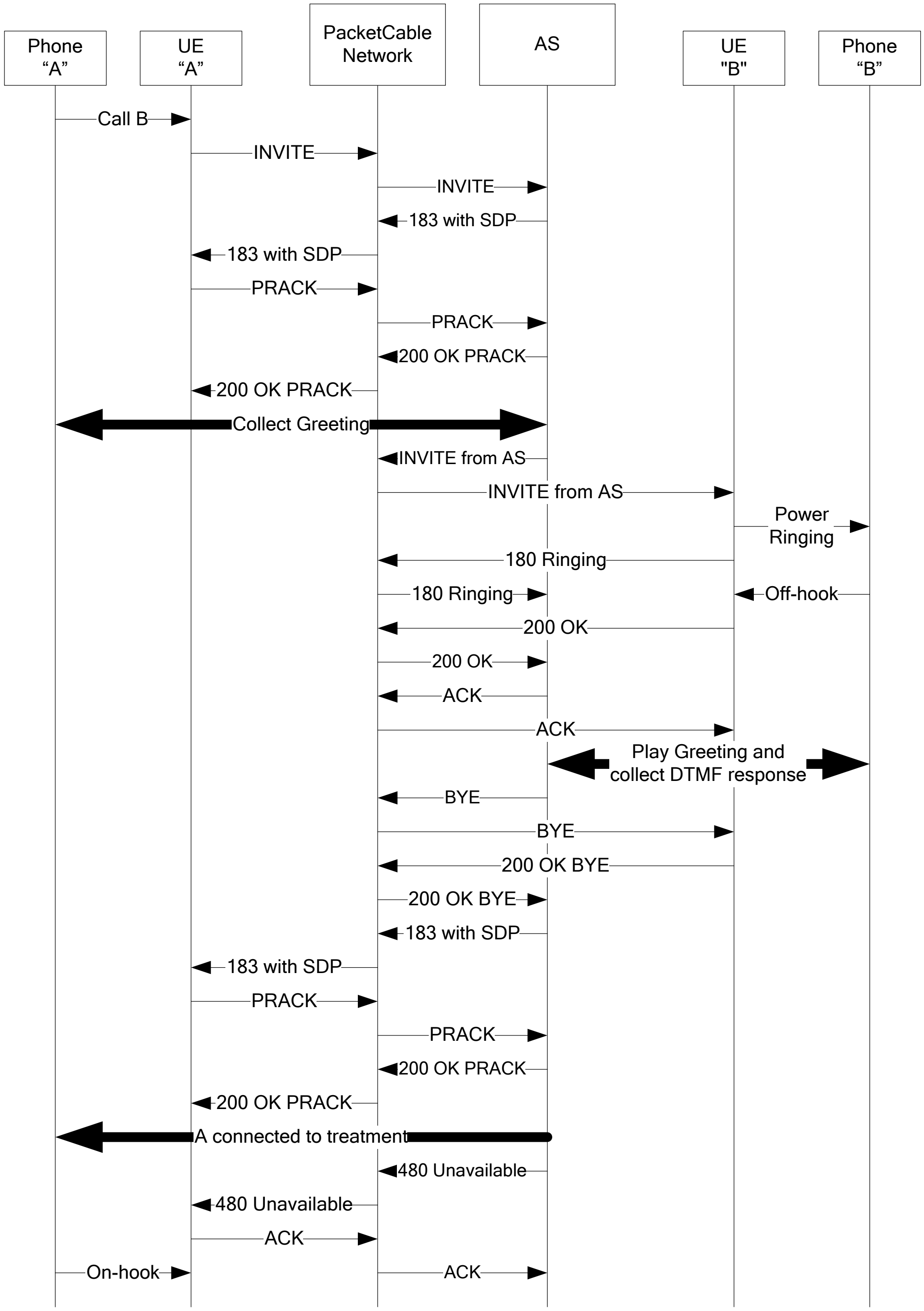

Figure 40 - SB; Called Party Rejects the Caller or Fails to Respond to the IVR

Figure 41 illustrates a call flow in which the caller fails to enter a greeting.

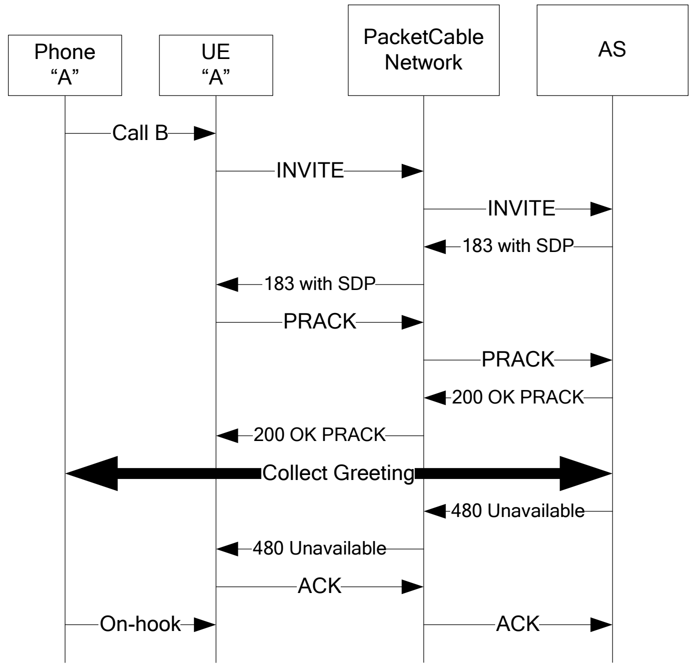

Figure 41 - SB; Calling Party Fails to Enter a Greeting

7.5 Multi-Party Features

7.5.1 Hook Flash Processing

The services defined within Section 7.5 all make use of user interaction via hook flash to trigger service logic. A user performs a hook flash by quickly depressing the button or "hook" that a traditional telephone handset rests upon when not in use, and then letting the hook rise again. The following sub-sections define the required behavior of the UE when processing the hook flash action.

The following requirements define the handling of an initial hook flash. When multiple hook flashes are handled through the invocation of a service, the behavior for each subsequent hook flash is defined within the service specific section.

- On detection of a hook flash, the UE MUST hold the active media stream, as defined in Section 7.1.5;
- If there is only a single dialog at the UE, then the UE MUST provide a recall dial tone and begin digit collection;
- If the service logic requires digit collection and the long inter-digit timer expires, the media MUST be retrieved, as defined in Section 7.1.5. Following retrieval of the media, silence, and then a ROH (Receiver Off-Hook) tone MUST be provided by the UE;
- If the subscriber flashes subsequent to an initial hook flash and prior to the completion of digit collection, the UE MUST abandon the second call leg attempt and retrieve the initial call leg media as defined in Section 7.1.5;
- If the subscriber hangs up when only one media stream exists, and that media is held as a result of a hook flash, then the UE MUST NOT send a BYE to the network. Instead, the UE MUST apply power ringing. If the subscriber goes off-hook before expiration of the ringing timer, the held media stream is retrieved as

defined in Section 7.1.5. Otherwise, on expiration of the ringing timer, the UE MUST release the call by sending a BYE.

7.5.2 Call Waiting

The Call Waiting (CW) feature provides the subscriber with the ability to detect an incoming call while in an existing call, and to answer that call by flashing, putting the existing call on hold. The behavior is described by Telcordia [GR 571] and [GR 572], unless otherwise specified here. There are two parts to this feature: activation/deactivation (disabling and enabling on a per call basis), and execution.

7.5.2.1 Feature Description

Service disabling and enabling occur when the subscriber calls a toggling vertical service code (for example, *70). The subscriber hears a tone indicating the state of the service (enabled or disabled) or a treatment if the service is not available for control (not assigned). Service control should be allowed any time the subscriber is allowed to place a call. Service control only applies to an existing call or to a new call (that is, from the time the caller initiates the control action until the time the caller hangs up).

7.5.2.2 Feature Activation and Deactivation

To enable and disable call waiting, the subscriber dials the VSC as normal call origination or after the flash in a second call. The UE recognizes the Call Waiting activation code as defined by its digit map and transformation table and invokes the CW-TOGGLE digit map procedure.

As part of its CW-TOGGLE digit map procedure, the UE MUST check the value of the Per-Call CW status in the UE. If the CW feature is active for the subscriber and if the Per-Call CW status is enabled, the UE MUST set the status to disabled in its CW-TOGGLE digit map procedure. If the Per-Call CW status is disabled, the UE MUST set the status to enabled in its CW-TOGGLE digit map procedure.

The UE MUST play a confirmation tone upon successful execution, play a recall dialtone, and begin to collect digits for call origination according to the default digit map, in its CW-TOGGLE digit map procedure. The UE MUST play a busy tone if the feature is inactive for the subscriber, or upon failure, in its CW-TOGGLE digit map procedure.

7.5.2.3 Feature Execution

Service execution consists of call termination (similar to basic call termination) while the UE is in an established call. The UE MUST reject the incoming INVITE with a 486 Busy Here final response if the UE is already engaged in an active call that cannot be held due to any of the following resource or service precedence restrictions:

- There is already a call on hold.
- The current call is an emergency call.
- The current call is a three-way call.

If there is an active call that can be held when another INVITE is received by the UE, the UE MUST play a call-waiting tone in the handset. The tone SHOULD match the agreed cadence for the alert-info header if such a header is present in the INVITE of the incoming call. Otherwise, the tone MUST be as specified in [GR 571]. The terminating UE MUST play the tone so that only the terminating subscriber hears the call waiting tone.

If the UE detects a flash, it MUST initiate a hold operation on the existing call's session, as defined in Section 7.1.5. The UE MUST accept the new call from the network once the re-INVITE for the old call completes, and stop playing the call waiting tone to the handset. The UE MUST reject the incoming call with a 486 Busy Here final response if the hold attempt of the existing active call fails.

If the UE detects a disconnect, it MUST release the existing call by sending a BYE to the network. The UE SHOULD proceed with normal SIP signaling for call termination on the waiting call. The UE MUST play ringtone, and the ringtone MUST match the agreed cadence for the alert-info header if such a header is present in the INVITE of the incoming call.

If the UE does not detect a flash or disconnect, it does nothing unusual from the SIP signaling point of view, but MUST stop playing the call waiting tone after applying the second tone to the handset for the duration and repetition interval, as defined in [GR 571].

7.5.2.4 Feature Data

Table 30 summarizes the feature data defined to support implementation of call waiting.

Table 30 - Call Waiting Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Feature Availability Status	Boolean	Non-volatile on XDS, Volatile on UE	Per public user ID	XDS	XDS	UE	Mandatory
Per Call CW disable star code	Digit map	Non-volatile on XDS, Volatile on UE	Per public user ID	XDS	XDS	UE	Mandatory
Feature Activation status	Boolean	Volatile	Per public user ID	UE	UE	UE	None

7.5.2.5 Feature Interactions

The call waiting feature has the following interactions.

Basic Call

The UE MUST respond with a 486 Busy Here final response when a new INVITE is received and the UE is off hook but is not yet in a stable two-way call (for example, while dialing). Further, the UE MUST respond with a 486 Busy Here final response when a new INVITE is received and when the UE is on hook but has not yet sent a BYE (see on-hook processing, Section 7.1.4.3).

Distinctive Alerting

Distinctive Alerting may be applied for a call in wait, as per Section 7.6.3.

DND, network

The interaction is consistent with basic call termination.

Call Forwarding

Selective Call Forwarding and Call Forwarding Variable take precedence over Call Waiting.

Call Hold

Call hold takes precedence over call waiting.

Call Transfer

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

When the subscriber executes a hook flash while a call is waiting, the UE MUST answer the waiting call instead of performing a call transfer.

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

Three-way Call

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

Emergency Call Origination

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

Solicitor Blocking

Solicitor Blocking takes precedence over Call Waiting. Since Solicitor Blocking is performed by an AS, and Call Waiting is performed by the destination UE, Solicitor Blocking always takes precedence over Call Waiting, because the AS receives the call before offering the call to the UE.

7.5.2.6 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

Figure 42 illustrates answering a waiting call with the call waiting feature. The UE receives an INVITE for a new call from the network. The UE sends a provisional response while proceeding to hold the existing call via re-INVITE. The UE then accepts the new call after holding the existing call.

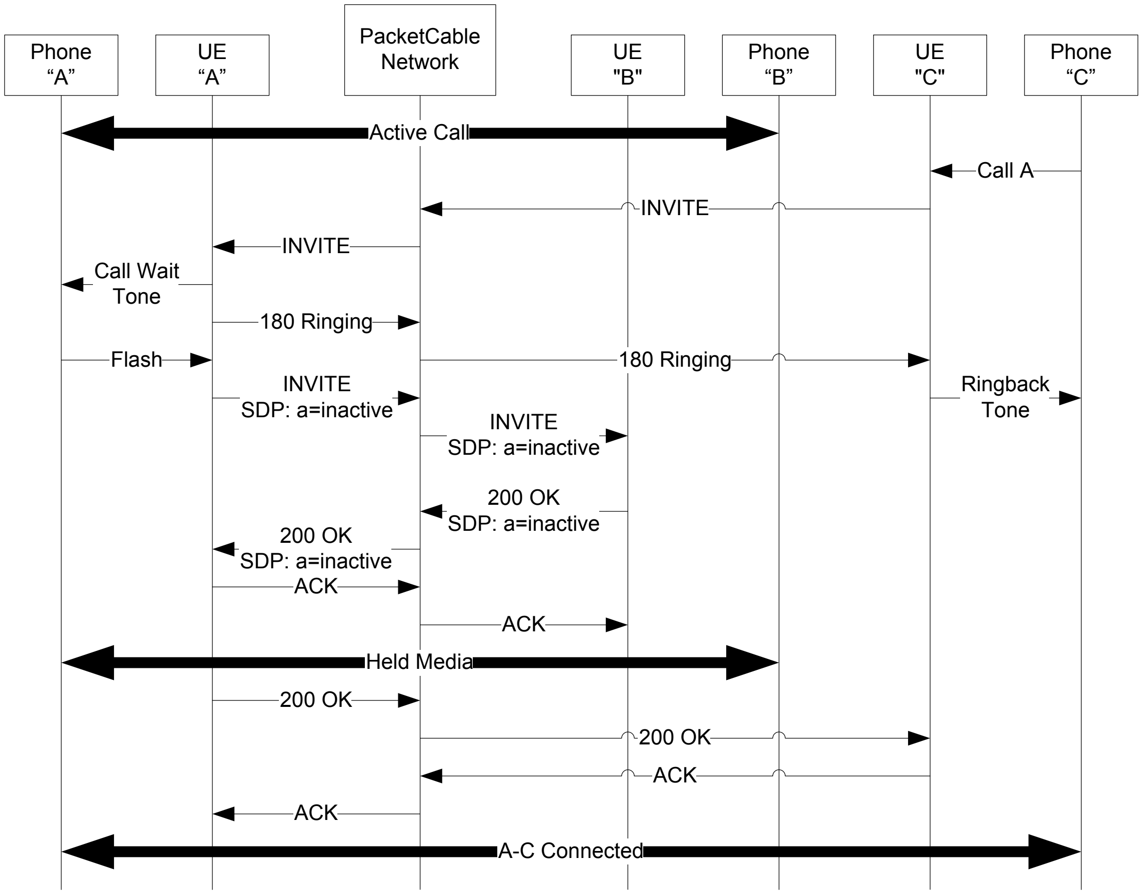

Figure 42 - Call Waiting (CWT); Answering a Second Call

7.5.3 Hold

7.5.3.1 Feature Description

The Call Hold (CHD) feature allows a user to put an active call on hold with a VSC for an extended period, while not sending or receiving any media to or from the held party, and then to retrieve the call at a later time. During the time that a call is on hold, the UE can place another call. If the user places another call while having a call on hold, the user is able to alternate the active call and the call on-hold using the same vertical service code.

A UE that supports the CHD feature MUST comply with the CHD requirements in [GR 579].

There are two types of CHD: silent on-hold and music on-hold. Silent on-hold is specified in the following sub-sections. Music on-hold is out of the scope of this specification. In the following sub-sections, CHD is assumed to mean silent on-hold.

7.5.3.2 Feature Activation and Deactivation

To activate the CHD feature, the user presses hook flash, hears a recall dial tone, then dials the CHD vertical service code. The UE recognizes the CHD vertical service code as defined by its digit map and transformation Table and invokes the HOLD-ACTIVATE digit map procedure.

The HOLD-ACTIVATE digit map procedure checks the value of the HOLD-ACTIVATE state in the UE. The state has two values: Active, indicating a call is on hold and Inactive, indicating no call is on hold.

Given the HOLD-ACTIVATE state of Inactive, the UE MUST play the recall dialtone, and begin collecting digits according to the default digit map, in its HOLD-ACTIVATE procedure.

Given the HOLD-ACTIVATE state of Active, the UE MUST retrieve the held call, as described in Section 7.5.3.3, in its HOLD-ACTIVATE procedure.

7.5.3.3 Feature Execution

The UE MUST hold the media (see Section 7.1.5). The UE MAY play comfort noise or silence insertion packets to and from the user while the media is being held.

When the CHD vertical service code is dialed after the UE receives a hookswitch flash, then the UE MUST apply the recall dial tone to allow the user to place another call.

If the short digit timer expires after the recall dial tone is applied, and no digits are dialed, then the UE MUST apply silence until a provisioned timeout. Once this provisioned timeout has expired, the UE MUST apply the ROH tone. If the user flashes and enters the CHD vertical service code, then the UE MUST take the held call off hold as defined within Section 7.1.5.

If the new call attempt fails after dialing at least one digit and before the new call attempt becomes a stable call, the UE MUST play silence to the user. If the user executes a flash hook, the UE MUST retrieve the held call without requiring that the vertical service code be dialed. This is the only situation in which the user can retrieve the held call without dialing the vertical service code.

If the UE receives a BYE on the held call's dialog while the user is making another call attempt, the UE MUST terminate the held call by sending a 200 OK for the BYE. The UE MUST then allow the second call attempt to continue.

Once the new call attempt is successful, the UE MUST allow the user to alternate the active and held calls by hook flashing, receiving a recall dial tone, and entering the vertical service code. A held call is retrieved as defined within Section 7.1.5.

If the UE detects on-hook while a call is on-hold, the UE MUST terminate the active call. The UE MUST apply ringing to alert the user that there is a held call. If the user doesn't answer after the ringing timer expires, the user's UE MUST send a BYE to end the session. When the held party's UE receives a BYE while on hold, the held party's UE MUST play the reorder tone to the local user.

If the non-held, non-controlling user disconnects, the disconnecting party's UE MUST send a BYE to the controller's UE. On reception of a BYE in this instance, the controller's UE MUST conform to the requirements for extended off-hook behavior, as defined in Section 7.1.4.4.

The UE MUST set its HOLD-ACTIVATE state to inactive once the held call is retrieved or disconnected. The UE MUST set its HOLD-ACTIVATE state to active once a call is successfully held.

The UE MUST allow only one call to be held at a time.

7.5.3.4 Feature Data

Table 31 summarizes the feature data that are defined to support implementation of Call Hold.

Table 31 - Call Hold Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Feature Availability Status	Boolean	Non-volatile	Per public user ID	XDS	XDS	UE	Mandatory
Feature code	Digits	Non-volatile	Per public user ID	XDS	XDS	UE	Mandatory
Feature Activation / Deactivation Confirmation Indicator	Tone or announcement	Non-volatile	Per public user ID	XDS	XDS	UE	Mandatory

7.5.3.5 Feature Interaction

Basic Call

The UE MUST be able to initiate an outgoing call while having a call on hold.

If the UE already has a call on hold when it receives a new INVITE, it MUST reject the incoming call with a 486 Busy Here response.

Transfer Call

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

The UE MUST NOT be able to transfer the held party and the non-held party to each other in either direction.

Conference Call

The UE MUST NOT permit a conference call between the non-held and held calls.

The UE MUST permit conference calls between non-held parties.

Call Waiting

If the UE has a held call and an active call at the same time, it MUST NOT provide call waiting, because hook flash is used to retrieve the held call.

Emergency Calling

A UE MUST NOT place emergency calls on hold.

7.5.3.6 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

The upper half of the call flow in Figure 43 is for UE A to put a party (UE B) on hold. The bottom half of the call flow is for UE A to retrieve the held party (UE B).

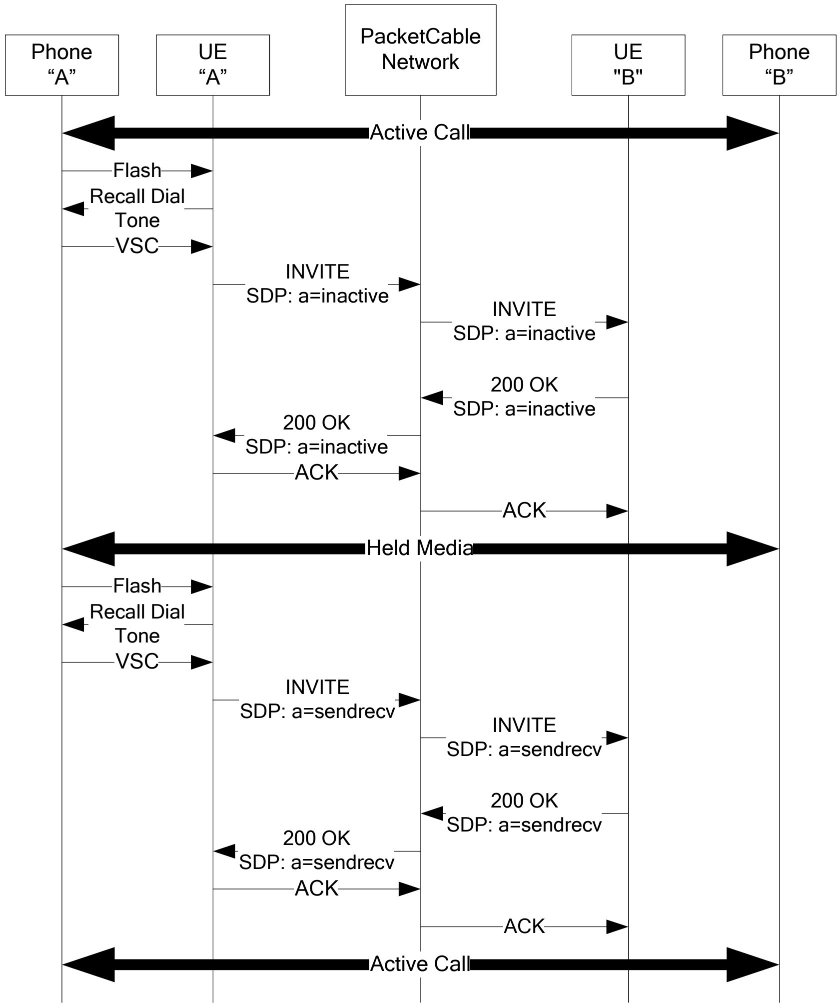

Figure 43 - Network Hold; Hold and Retrieve

Figure 44 illustrates network hold, in which the second call attempt fails.

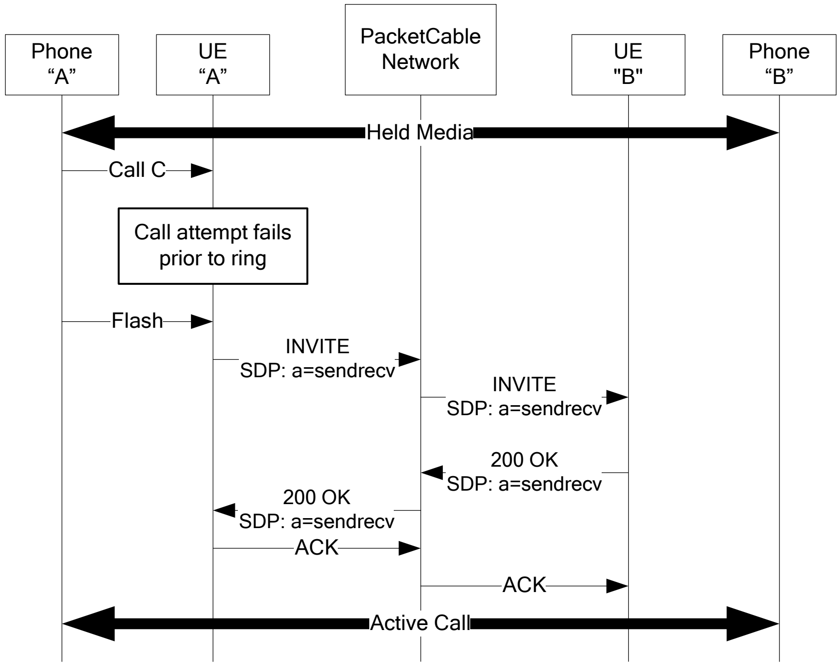

Figure 44 - Network Hold; Second Call Attempt Failed

Figure 45 illustrates a network hold scenario in which the controller goes on-hook.

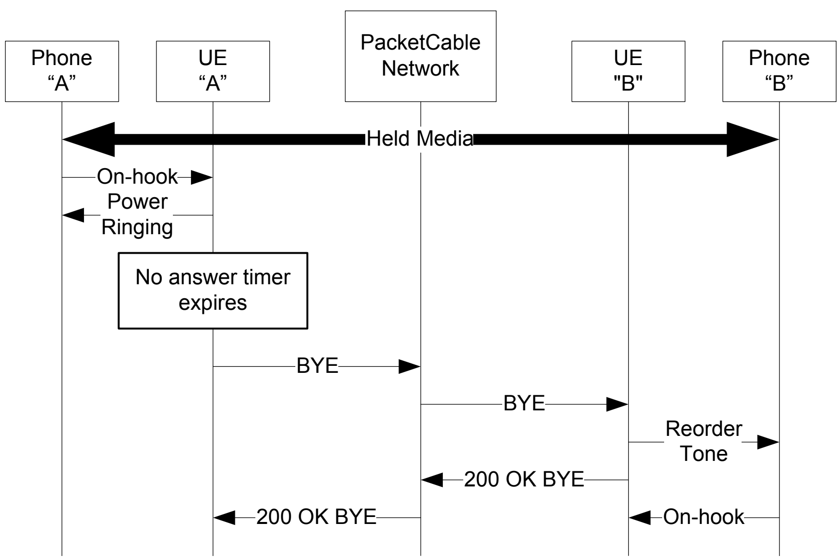

Figure 45 - Network Hold; Controller Goes On-Hook

7.5.4 Call Transfer

This section specifies the transfer of one caller (the transferee) to another party (the transfer-to party), in which the transfer is initiated by the originator of the second call (the transferor). As described in the [GR 579], there are two common varieties of call transfer - consultative and blind. Consultative transfer is specified by [GR 579]. Blind transfer is not explicitly specified by Telcordia as it is with numerous implementations in Centrex systems, which have the benefit of additional inputs (such as transfer buttons) or vertical service codes for feature control. This specification defines blind transfer in terms of the state of the second call: if the call is unanswered before completion of the transfer sequence, the transfer is considered blind.

7.5.4.1 Feature Activation

Call Transfer (CT) is always active once assigned to the subscriber. Once the service is assigned, a UE MUST include an Allow header containing REFER in every request in which an Allow header is allowed, according to [RFC 3261].

7.5.4.2 Feature Execution Overview, Preconditions

The UE MUST only begin transfer execution if there is one active call and there is no other call present or pending (waiting). Figure 46 describes execution of this feature.

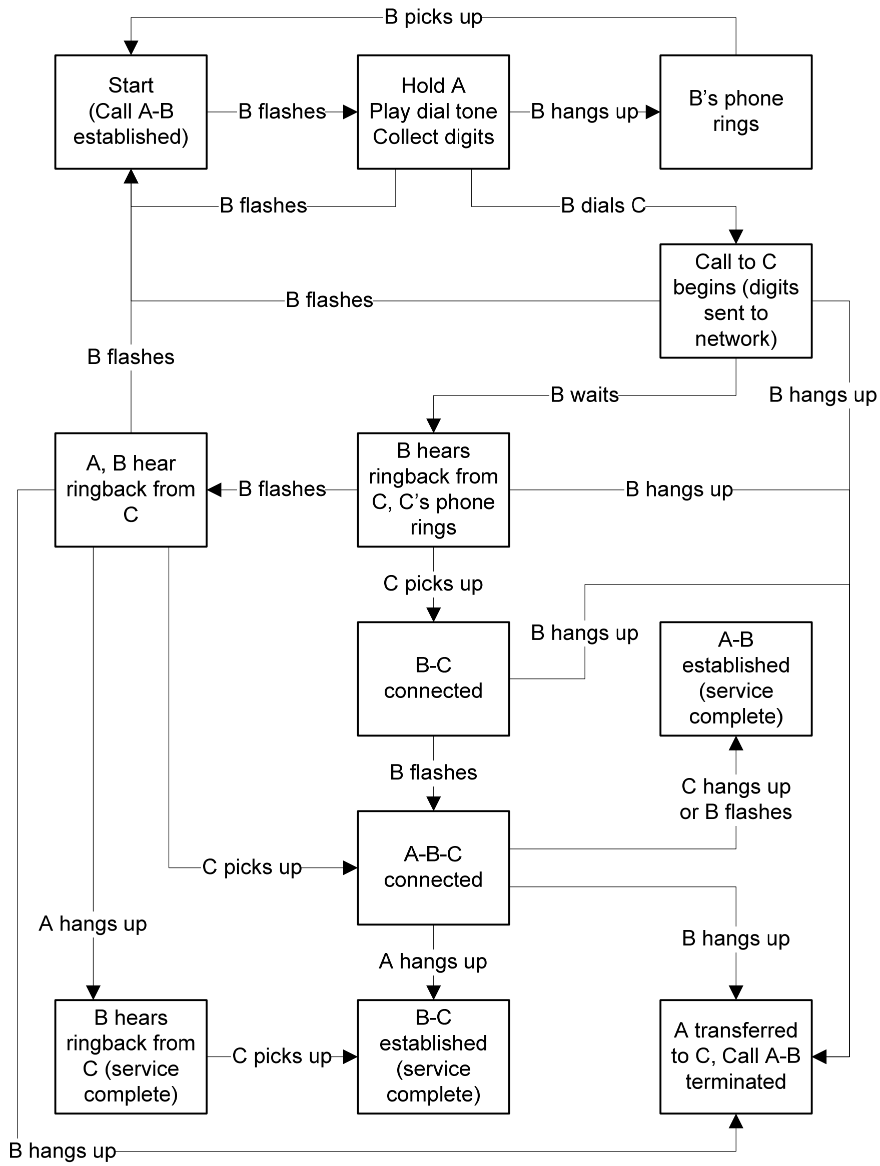

Figure 46 - Call Transfer (CXR) Feature Flow; B Transfers A to C; Not All User Inputs Are Shown for A

Transfer is initiated via a REFER sent out of dialog, as described in [RFC 3515], to the transferee. The UE MUST include a Refer-to header in the REFER to indicate the call legs being connected. The transferor's UE MUST include a nested Replaces header in the Refer-to header to direct the transferee to the transfer-to party. The transferor's UE MUST set the Refer-to header to contain a nested Replaces header (as defined by [RFC 3891]) identifying the dialog to replace. The transferor's UE MUST include the Target-Dialog header [RFC 4538] which describes the existing dialog between the transferor and the transferee. If the transferee does not support the Target-Dialog header (indicated by the dialog options tag), the transferor MUST NOT send the REFER to the transferee to initiate transfer.

In order to preserve the privacy of the parties involved and deter fraudulent use of this feature, the transferor's S-CSCF should be provisioned by the service provider with an initial filter criterion that sends the REFER to an AS (the CT AS).

The CT AS receiving the REFER MUST cache the contents of the Refer-to and replace the contents of the Refer-to with its private URI. The CT AS MUST proxy the REFER to the intended target.

Upon receiving the subsequent INVITE from the transferee, the CT AS MUST verify the originator matches the destination of a previously proxied REFER, replace the contents of the request URI with the cached Refer-to, add the Replaces header from the cached Refer-to, and proxy the INVITE accordingly.

The UE receiving the REFER proxied by the CT AS MUST support the Refer-to header, and send subsequent NOTIFYs to the transferor to indicate transfer status. The MGC requirements for support of REFER method and Refer-to header for Call Transfer are specified in [CMSS1.5]. An MGC receiving the REFER proxied by the CT AS supports the Refer-to header. The MGC sends subsequent NOTIFYs to the transferor to indicate transfer status.

7.5.4.3 Feature Execution, Transferor

The UE MUST ignore a hookflash if the call transfer feature is not enabled for the UE in PACM (via the Boolean Transferor Feature Availability Status).

If the call transfer feature is enabled for the UE in PACM (via the Boolean Transferor Feature Availability Status), the UE MUST send a re-INVITE to hold the remote party upon detection of a hookflash. The UE MUST then play dial-tone and collect digits for second call origination. The UE MUST abandon the second call attempt if the subscriber flashes and return to the first call if digit collection is not completed successfully.

Upon detection of a valid digit string, the UE MUST initiate the second call via an INVITE. The UE MUST apply all features to the second call as with normal call origination.

The transferor's UE MUST mix the RTP streams of the transfer-to party and the transferor to be played back to the transferee (the calls are conferenced locally on the transferor's UE) if the subscriber flashes after second call initiation.

When the subscriber hangs up, the transferor's UE MUST send a REFER to the transferee on a new dialog, having a Refer-To header containing a nested Replaces header with the transfer-to the party's SIP dialog information and a Target-dialog header matching the transferee's existing SIP dialog with the transferor. The transferor's S-CSCF is assumed to be provisioned with an initial filter criterion to send the REFER to the CT AS.

The transferor's UE SHOULD wait for NOTIFY message(s) from the final recipient of the REFER indicating the transfer result.

If the transferor hangs up before the transfer-to party picks up, the transferor's UE MUST send CANCEL to the transfer-to party and BYE to the transferee upon one of the following conditions:

- Receipt of a NOTIFY message from the final recipient of the REFER indicating successful initiation of the transfer.
- Expiration of the Notify-timeout before receiving a NOTIFY from the final recipient of the REFER

If the transferor hangs up after the transfer-to party has picked up, the transferor's UE MUST send BYE to the transferee and the transfer-to party upon one of the following conditions:

- Receiving a NOTIFY message from the final recipient of the REFER indicating successful initiation of the transfer.
- Expiration of the Notify-timeout before receiving a NOTIFY from the final recipient of the REFER

7.5.4.4 Feature Execution, CT AS

When the CT AS receives a REFER from the transferor, the CT AS MUST:

- Validate the subscriber's ability to transfer, and reject if invalid.
- Replace the Refer-to header content with its private URL.
- Proxy the REFER to the destination indicated in the request URI.
- Add its record route entry to the REFER before proxying in order to receive subsequent NOTIFY messages from the recipient of the REFER.
- Proxy NOTIFY messages to the S-CSCF.

When the CT AS receives an INVITE with Replaces header, the CT AS MUST:

- Match the INVITE to a previously proxied REFER, and reject if no match is found.
- Replace the request URI in the INVITE with the original Refer-to contents, inserting the embedded Replaces header as a discrete header within the INVITE.
- Proxy the INVITE.

7.5.4.5 Feature Execution, Transferee

When the transferee's UE receives an out-of-dialog REFER, it MUST:

- Match the Target-dialog in the REFER with an existing dialog for transfer.
- Reject the REFER with a 481 Call/Transaction Does Not Exist final response if no match is found between the Target-dialog header and an existing dialog.
- Send an INVITE using the Refer-To header in the REFER for the request URI
- Notify the transferor of call status with a NOTIFY.

7.5.4.6 Feature Execution, Transfer-to party

When the transfer-to party's UE receives an out-of-dialog INVITE, it MUST:

- Match the content of the Replaces header in the out-of-dialog INVITE to the existing dialog with the transferor. NOTE: Use of the replaces header on an early dialog between the transfer-to and transferee in the case of blind transfer diverges from the use specified in [RFC 3891].
- Reject the INVITE with a 481 Call/Transaction Does Not Exist final response if the Replaces header does not match the existing dialog with the transferor.

7.5.4.7 Feature Data

Table 32 summarizes the feature data items that are defined for the implementation of Call Transfer.

Table 32 - Call Transfer Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Feature Availability Status	Boolean	Non-Volatile	Per public user ID	XDS	XDS	UE	Mandatory
Notify-timeout	Integer (seconds)	Non-Volatile	Per public user ID	XDS	XDS	UE	Mandatory

7.5.4.8 Feature Interactions

Basic Call

The transferor's UE MUST use the same data as basic call (see Section 7.1) for the second call origination.

Call Waiting

If the subscriber's UE is engaged in an active call, a new INVITE is received at the subscriber's UE, and the subscriber has the Call Waiting feature, then, for as long as the UE is engaged in two dialogs, the subscriber's UE MUST NOT invoke the Call Transfer feature when the subscriber hook flashes. If the subscriber's UE receives a new INVITE at any point after the subscriber has hook flashed to initiate a transfer, and the subscriber has the Call Waiting feature, then the subscriber's UE MUST respond to the new INVITE with a 486 Busy Here response.

Three-Way Calling

Call Transfer takes precedence over three-way calling.

Call Blocking, Outbound (In General)

If the subscriber invokes the CT feature and receives recall dial tone and dials a blocked phone number in response to the recall dial tone, the OCB AS MUST block the call according to the procedures in Section 7.4.1. This assumes that the S-CSCF is correctly provisioned with initial filter criteria to trigger the INVITE to the OCB AS prior to triggering the call to the CT AS.

Emergency Calling

When the subscriber's UE is engaged in an active Emergency Call and the subscriber performs a hook flash, the subscriber's UE MUST NOT play the recall dial tone to the user as described in this CT section, but continues with the Emergency Call instead.

Caller ID Blocking

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

7.5.4.9 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

There are two relevant call flows for call transfer. The subscriber originates the second call, waits for ringback, and then hangs up, as shown in Figure 47.

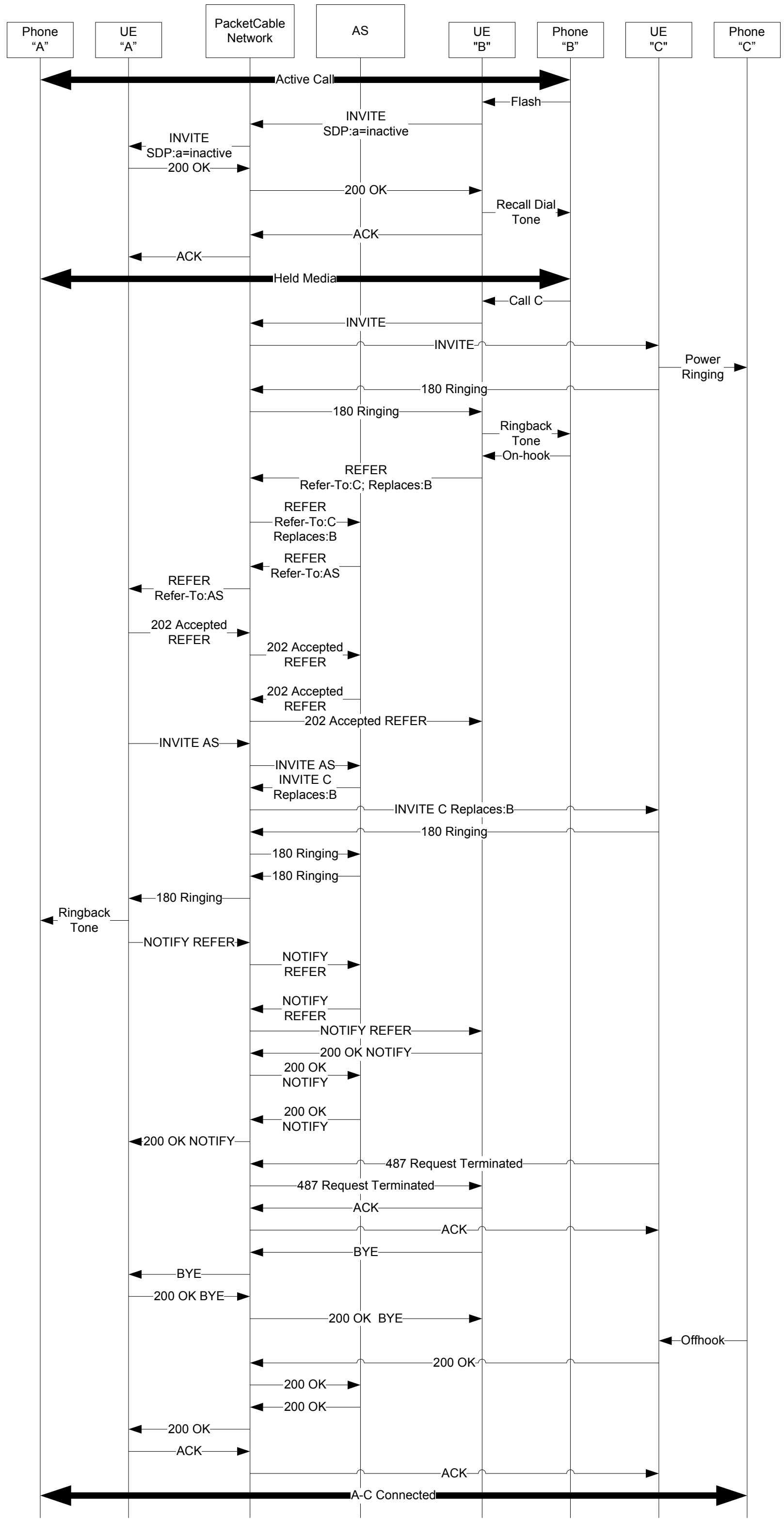

Figure 47 - CXR; B Transfers A to C (blind) Using REFER after Initial INVITE but Before C Answers

The subscriber originates the second call, waits for an answer, and then hangs up, as shown in Figure 48.

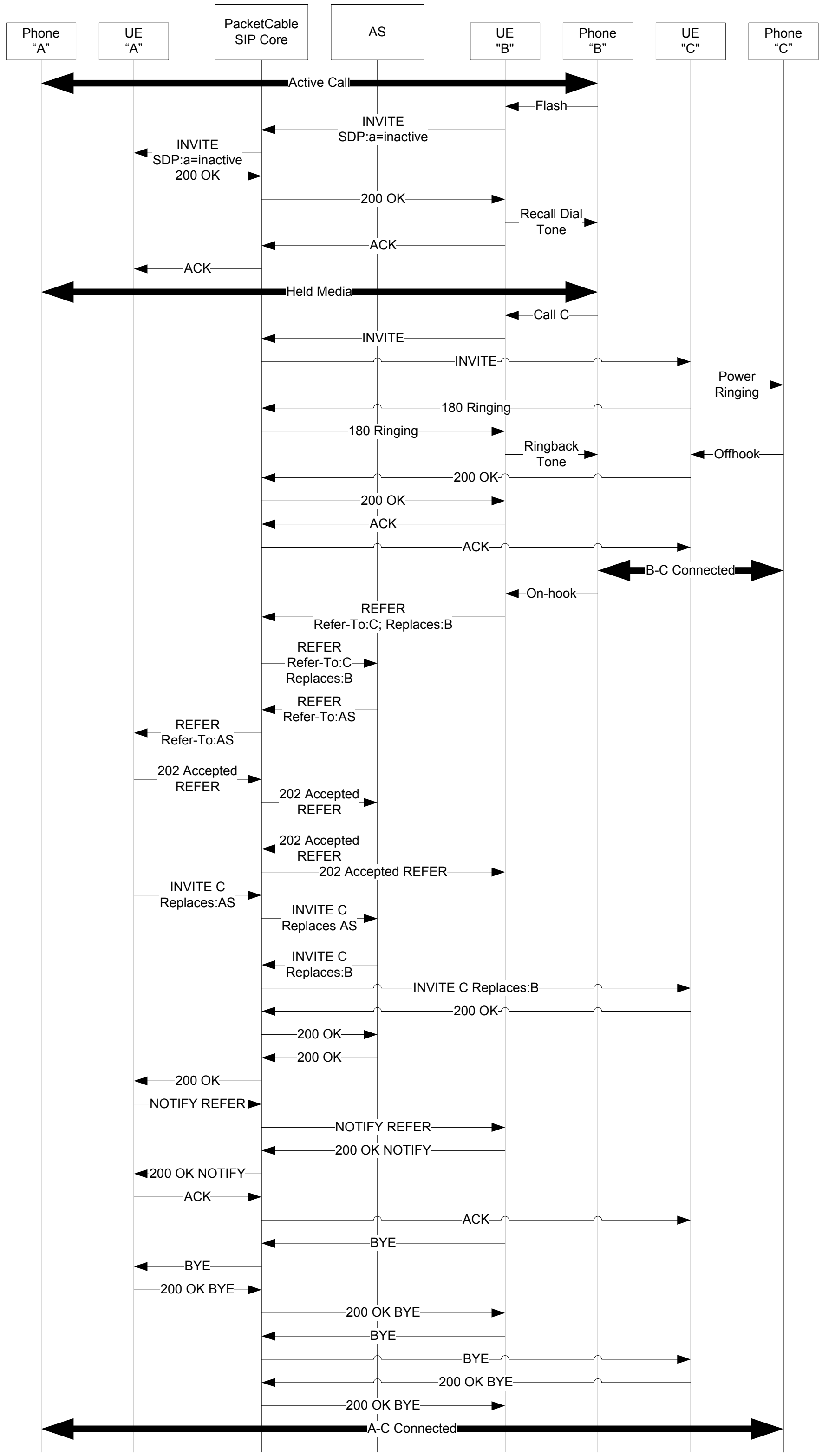

Figure 48 - CXR; B Transfers A to C (Consultative) Using REFER after Establishing the Call to C with INVITE

7.5.5 Three-Way Calling

7.5.5.1 Feature Description

Three-Way Calling (3WC) allows a subscriber to add a third party to an existing two-party active call without operator assistance. The subscriber initiating the 3WC is called the controller. A UE with an analog user interface for 3WC MUST comply with [GR 577].

7.5.5.2 Feature Activation and Deactivation

3WC feature availability is provisioned via PACM. UEs not initiating a 3WC do not need to be provisioned with the 3WC feature in order to participate in a 3WC.

While a subscriber is in an active two-way call, the subscriber activates the 3WC feature by hook switch flash, then dials the destination address. After the destination is ringing or answered, the subscriber can hook flash again to join the held with the third party forming a 3WC. The controller can hook flash again to disconnect the third party from the 3WC. The controller and the remaining party form a Two-Way Call. If the controller hook flashes again, the UE is treated as if it is initiating another 3WC.

To deactivate the feature, the controller either hook flashes to disconnect the third party or hangs up while a 3WC is active.

7.5.5.3 Feature Execution

Be definition, before a UE can start a three-way call, the UE is assumed to be involved in a stable call. It does not matter which party originated the call.

Upon detection of a hook flash, the controlling UE MUST place the first call on hold (see Section 7.1.5). Once the first call is on hold, the UE MUST apply the provisioned recall dial tone and be prepared to collect digits for a second call leg.

After receiving a recall dial tone, the subscriber dials the address of the third party. The UE MUST initiate a call on receiving a valid dialed digit string. If the second call attempt fails, the held call can be retrieved by the controlling UE via a subsequent hook flash.

If the held call is disconnected while the controller is making the second call attempt, the held call MUST be terminated. After the held call is terminated the second call attempt MUST continue as a two-party call.

If the second call attempt succeeds, the controller may speak to the third party privately. Otherwise, the controller can join the two call legs together after the ringing is applied and before the third party answers the call. The subscriber hook flashes again to join the held call with the second call leg to form a Three-Way Call (3WC). If a UE has a held call and an active call and detects a hook flash, the UE MUST create a conference by mixing the audio received on the two call legs with the audio received from the local party, and send the mixed audio in the RTP streams sent to the two call legs.

If the controlling UE detects a hook flash during a 3WC, the UE MUST send a BYE to disconnect the third party from the call. After the third party is disconnected, the controlling UE MUST cease mixing the audio and continue a two-way call with the remaining party.

If the controller's UE detects on-hook while having a call on-hold, before the third party answers, the controller's UE MUST terminate the call with the third party by sending a CANCEL. If the controller's UE detects on-hook while having a call on-hold, after the third party answers, the controller's UE MUST terminate the call with the third party by sending a BYE. In either case, after the controller's UE detects on-hook, the UE MUST apply ringing to alert the controller. If the UE does not detect off-hook prior to the expiration of the ringing timer, the UE MUST send a BYE to the second party to end the session.

If either the second or third party disconnects after a 3WC is established, the remaining call party's UE MUST continue in a two-way call.

If the controller hangs up the phone in an active 3WC, and the controller of the 3WC has not subscribed to the Call Transfer feature (see Section 7.5.4), the controller's UE MUST send BYE to the second and third parties to terminate each of these call legs. See the feature interactions section below for procedures for interactions between 3WC and Call Transfer.

The second or third party in a 3WC can hook flash to put the 3WC on hold and make another call, subject to the features to which they are subscribed. The remaining parties' UEs in the 3WC MUST continue the two-way media stream without being impacted.

7.5.5.4 Feature Data

Table 33 summarizes the feature data defined to support implementation of the three way calling feature.

Table 33 - 3WC Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Feature Availability Status	Boolean	Non-volatile	Per public user ID	XDS	XDS	UE	Mandatory

7.5.5.5 Feature Interactions

Basic Call

Once a controller's UE is in a 3WC and detects a hook flash, it MUST NOT give the user the ability to initiate another basic call.

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

Call Waiting

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

Call Transfer

When a controlling UE is in a 3WC, it MUST NOT perform Call Transfer by a hook flash, as this disconnects the third party in the 3WC.

A controlling UE of a 3WC that has the Call Transfer feature enabled MUST transfer when it detects on-hook.

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

Call Hold

Call hold takes precedence over three-way call.

Operator Service

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

Emergency Call

A UE MUST NOT place an emergency call on hold when it detects a hook flash. When a party is in a two-party call, and the 3WC feature is enabled, the UE puts the first party on hold when it detects a hook flash and then it begins digit collection. The UE establishes a session with the PSAP if an emergency call number is dialed. After the emergency call is answered, the UE MAY join the emergency operator and the held party to form a 3WC if it detects a hook flash. If the UE detects on hook before the emergency call is answered, the UE MUST send a BYE to the PSAP to release the session. Once the 3WC is established, the UE MUST ignore any detected hook flashes. While in a 3WC with an emergency operator, the UE MUST adhere to the emergency call procedures defined in Section 8.

7.5.5.6 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

Figure 49 shows the call flows for three parties to establish a 3WC.

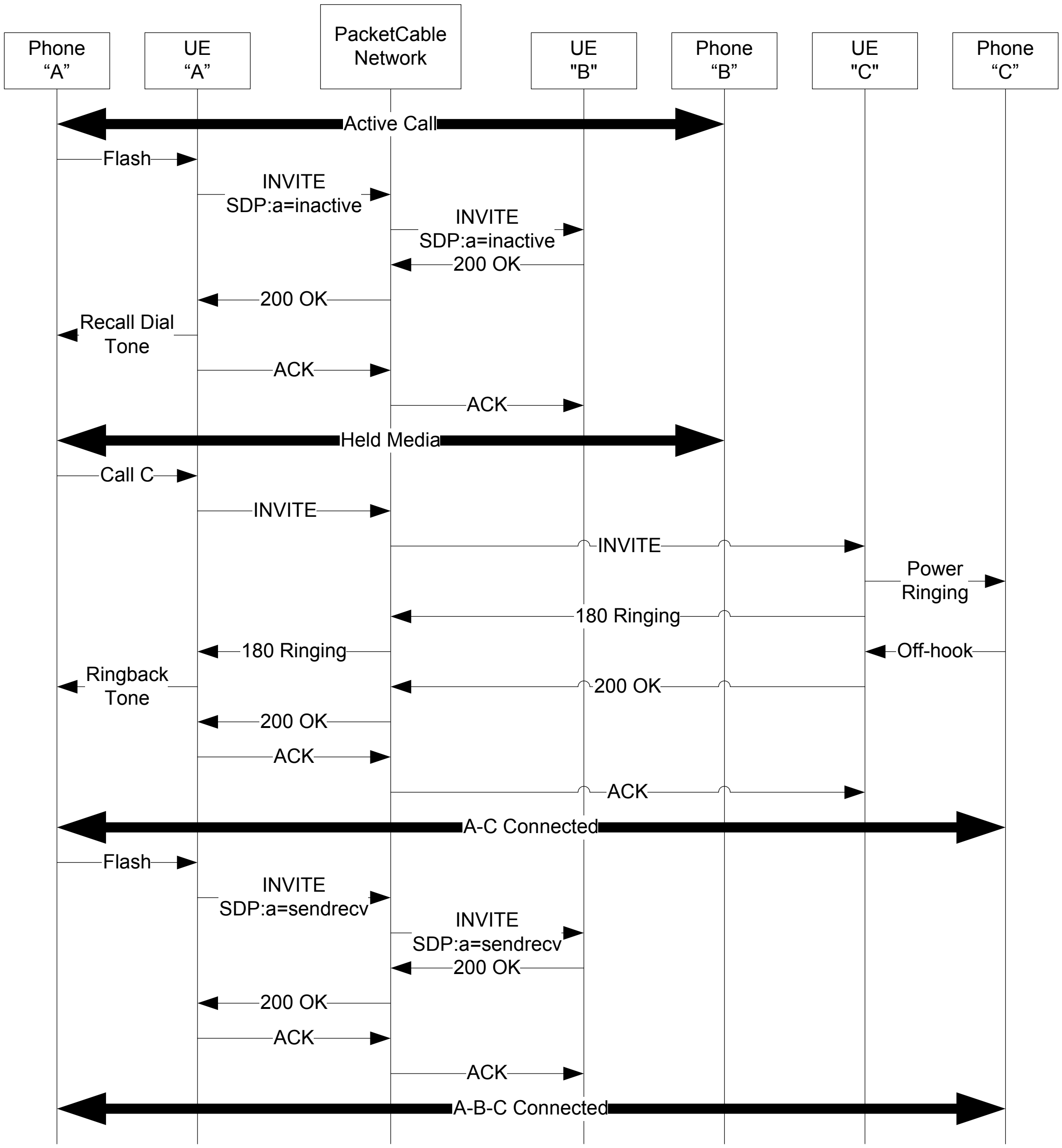

Figure 49 - Three-Way Call (3WC)

After the second call leg is successfully made, if the controller hangs up before joining the held call, the active call with the third party is terminated. The controller receives a ringing alert. If the controller goes off-hook again before the ringing timer expires, the controller receives a dial tone. The controller can hook flash to retrieve the held call, or the controller can dial digits again to make another second call attempt to make another 3WC. This is shown in Figure 50.

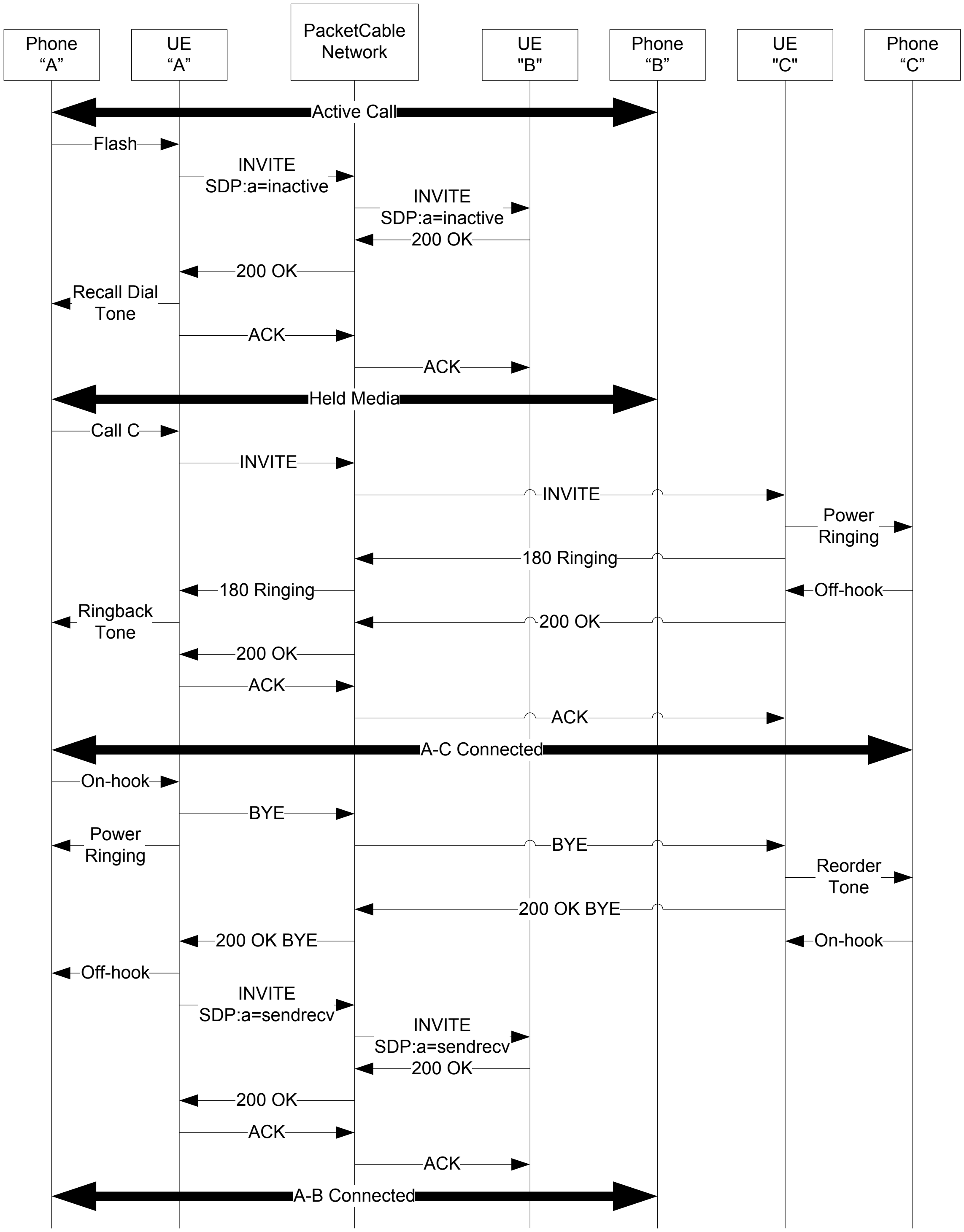

Figure 50 - 3WC; Controller hangs up during 2nd call attempt

If the controller hook flashes after establishing the 3WC, the third party that joins the call is disconnected. The other two parties continue in a Two-Way Call. This is shown in Figure 51.

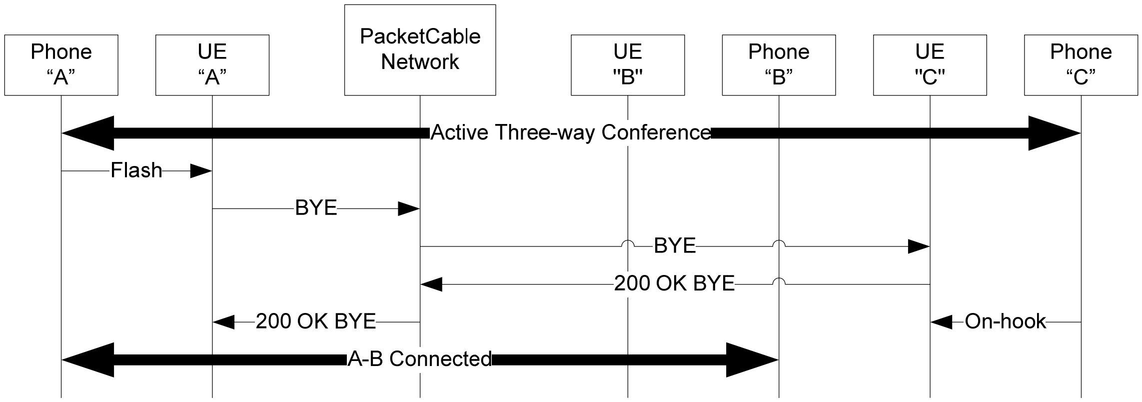

Figure 51 - 3WC; Controller Flashes the Hook to Disconnect Third Party in a 3WC

If the controller goes on hook during a 3WC, the conference call is disconnected. This is shown in Figure 52.

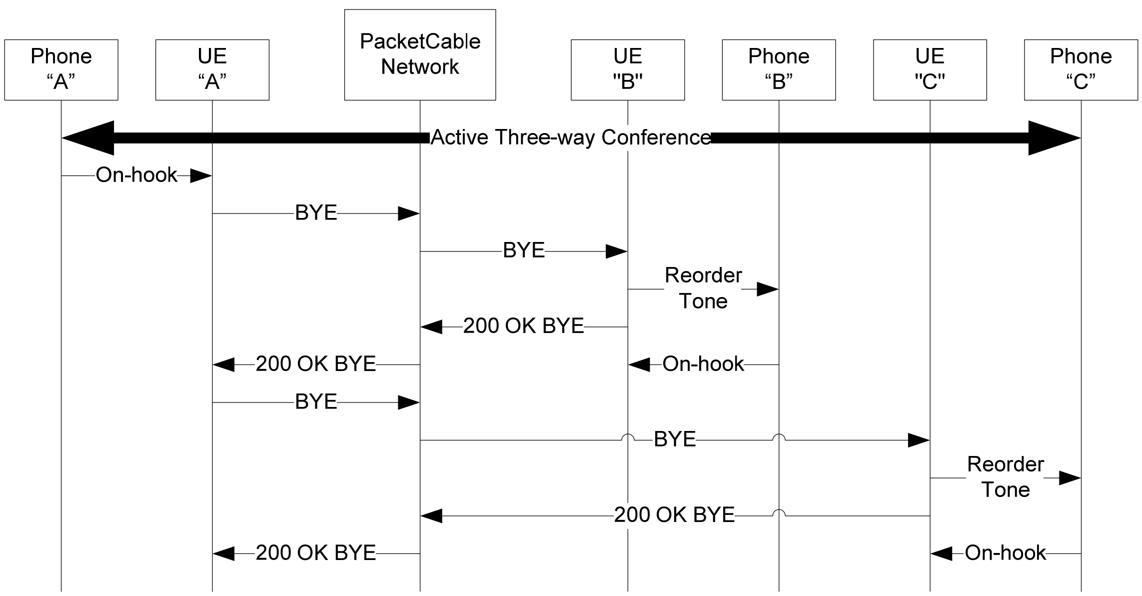

Figure 52 - 3WC; Controller Disconnects the 3WC

7.6 Miscellaneous Features

7.6.1 Do Not Disturb

7.6.1.1 Feature Description

The Do-Not-Disturb (DND) feature allows the user to put a UE into a mode in which the UE does not send an alert when a call is presented.

The user can activate or deactivate the DND feature by entering a Vertical Service Code (VSC). Once activated by the user, the DND remains in effect for a period of time (for example, three hours) as provisioned by the service provider. Alternatively, the user can schedule the DND activation and deactivation via a web portal provided by the service provider.

When the DND feature is in effect, the UE does not alert when a call is presented unless one of the following is true of the caller:

- The caller has a DND override privilege designated by the service provider,
- The caller is identified by the DND Subscriber as exempt from the application of DND treatment, or
- The caller possesses a DND override PIN.

DND is dependent upon the Screening List Editing (SLE) feature (see Section 7.6.7) for activating or deactivating the feature, modifying the list of callers who are exempt from DND handling, or obtaining a feature status report.

A subscriber performs these operations by going off-hook, receiving a dial-tone, and dialing the DND access code (typically *78 and/or *79). Historically, DND once had separate activation and deactivation VSCs, but Telcordia standards have changed recently and now support a single VSC that provides access to the SLE feature for updating the DND feature's configuration. In order to maintain backward compatibility with the older, two-VSC method for DND activation/deactivation, two VSCs may still be used, but either VSC invokes the same SLE feature for updating the DND feature's configuration. This allows operators who currently have customers using *78 and *79 as separate activation and deactivation codes to continue such use while permitting other operators who are not currently providing screening list features to advertise just one code as a single feature code.

Once either DND feature code is successfully entered, the customer receives announcements providing the following information (not necessarily in this order):

- The name of the service (that is, DND)
- The current status of DND (that is, active or inactive)
- The current size of the customer's DND list
- Actions and associated dialing codes available to the user:
    - Add entr(y)ies to the list
    - Delete entr(y)ies from the list
    - List review
    - Change status (that is, active to inactive or inactive to active).

7.6.1.2 Feature Activation and Deactivation

The user can activate or deactivate the DND feature via the telephony user interface (that is, the phone keypad) by entering a VSC. The DND VSCs are defined for the UE as part of the Digit Map as defined in Section 7.1.1. Upon matching the user-entered digits with the DND VSC, the UE carries out the DND-PROGRAM action. If the feature availability status indicates that the DND feature is not available, then the UE MUST NOT send the VSC to the DND AS. If the feature availability status indicates that the DND feature is available, then the UE MUST send the VSC to the DND AS via the mechanism specified in Section 7.1.2. The DND AS MAY, in turn, connect the user to an IVR system for DND programming and activation and deactivation.

As part of the DND programming, the DND AS MUST provide the user with an SLE feature, as described in Section 7.6.7, to program a list of callers (which includes their public user IDs) who are exempted from the DND treatment when the DND feature is in effect. This is the DND Exempt List defined in Table 34. The maximum length of the list SHOULD be provisionable by the service provider and have a default value of 12. As part of the data associated with the DND service defined in Table 34, the user has the ability to disable the use of the DND Exempt list and also to disable the use of PIN override. These capabilities are defined in Table 34 as "DND-Exempt-List Capability Status" and "DND-Override-PIN Capability Status" respectively.

Once the DND feature is activated, the DND AS MUST ensure that the feature remains in effect for a period of time provisioned by the service provider (the Do Not Disturb Subscription Duration specified in Table 34), with a default duration of three hours.

Upon activation or deactivation of the DND feature, the DND AS MUST send a NOTIFY for the RST Service Profile to the UE. The resulting NOTIFY MUST inform the UE about the new activation status, as defined in Table 35, using the mechanism specified in [RST PACM]. Upon receiving the NOTIFY message, the UE MUST present to the user the Feature Activation Confirmation Indicator or the Feature Deactivation Confirmation Indicator provisioned per Table 34.

Whenever the UE detects an off-hook condition and the DND feature is active (determined based on the feature activation status parameter as defined in Table 35), the UE SHOULD present the user with a recall dial tone (Table 2) to alert the subscriber that the DND feature is active.

The subscriber can activate or deactivate the DND feature via a web portal provided by the service provider. In this case, the web portal performs the necessary user authentication and makes the user-entered feature status available to the DND AS. The web portal can provide the subscriber with a tool to schedule DND sessions; however, such a tool is out of scope.

7.6.1.3 Feature Execution

Upon receipt of an INVITE, the DND AS associated with the callee decides if the INVITE needs to be presented to the called UE based on the current status of the DND feature. The current feature status is determined by the feature provisioning information and the activation/deactivation status.

If the DND feature is not active, the AS MUST forward the INVITE to the called party.

If the DND feature is active, the DND AS MUST determine if the caller has an emergency or operator services public identity and, therefore, has the automatic DND override privilege. If the caller has the automatic DND override privilege, the AS MUST forward the INVITE to the called UE.

If the DND feature is active, if the caller does not have the automatic DND override privilege, the DND-exempt-list capability is disabled by the subscriber, and the DND-override-PIN capability is disabled, the DND AS MUST play a DND announcement to the caller using the Caller Treatment Indicator, provisioned per Table 34, and terminate the call by sending a 403 Forbidden final response.

If the DND feature is active, if the caller does not have the automatic DND override privilege, and the DND-exempt-list capability is enabled by the user, the DND AS tries to match the calling party's public user ID with an entry in the DND-exempt-list. If a match is established, the DND AS MUST forward the INVITE to the called party. If a match is not found, and the DND-override-PIN capability is disabled by the user (reference Section 7.6.2), the DND AS MUST play a DND announcement to the caller using the Caller Treatment Indicator provisioned per Table 34, and terminate the call by sending a 403 Forbidden final response.

Otherwise, the DND AS MUST send back to the caller a reliable Session Progress (183) response with an early session disposition type, as defined in [RFC 3959], and appropriate SDP content for a two-way early media. The AS informs the caller that the called party is not available for the call and prompts the caller to enter a DND override PIN. The DND AS MUST validate the caller-entered PIN. If the PIN is correctly entered, the DND AS MUST forward the INVITE to the called UE. If the PIN is incorrectly entered (after a provisioned number of tries, the DND Override PIN Attempts in Table 34), the DND AS MUST terminate the call by sending a 403 Forbidden final response.

7.6.1.4 Feature Data

Table 34 summarizes the feature data items that are defined for the implementation of DND.

Table 34 - DND Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Feature Availability Status	Boolean	Non-volatile	Per public user ID	XDS	XDS	AS, UE	Mandatory
Feature Activation Status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
Caller Treatment Indicator	Strings representing treatment options	Non-volatile	Per public user ID	XDS	XDS	AS	None
DND Override PIN	Digits	Non-volatile	Per public user ID	AS	AS	AS	None
DND Override PIN Attempts	Integer	Non-volatile	Per public user ID	AS	AS	AS	None
Feature Activation Confirmation Indicator	Tone or announcement	Non-volatile	Per public user ID	XDS	XDS	UE	Mandatory
Feature Deactivation Confirmation Indicator	Tone or announcement	Non-volatile	Per public user ID	XDS	XDS	UE	Mandatory
DND Exempt List	List of public user IDs	Non-volatile	Per public user ID	AS	AS	AS	None
DND-Exempt-List Capability Status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
DND-Override-PIN Capability Status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
Do Not Disturb Subscription Duration	Integer (seconds)	Volatile on UE, Non-volatile on XDS	Per public user ID	AS	AS	AS	None

Table 35 - Do Not Disturb UA-Profile Event Package Notify Body

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Feature Activation Status	Boolean (0=inactive, 1=active)	Volatile on UE, non-Volatile on AS	Per public user ID	AS	AS	AS, UE	None

7.6.1.5 Feature Interactions

The feature has the following interactions.

Basic Calling Capabilities

UEs with DND activated MUST be able to originate calls.

UEs with both the DND and the MWI feature provisioned and active MUST NOT play on-hook message waiting audible indicator (ringsplash) if the MWI feature is configured for UE ringsplash when a message is waiting.

Multi-Party Features

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

Call History Features

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

Call Forwarding

DND takes precedence over Call Forwarding Variable.

DND takes precedence over Call Forwarding to Voice Mail.

DND takes precedence over Call Forwarding Don't Answer.

DND takes precedence over Call Forwarding Busy.

Selective Call Forwarding takes precedence over DND.

Do Not Disturb

Busy Line Verify takes precedence over Do Not Disturb.

Anonymous Call Rejection

DND takes precedence over Anonymous Call Rejection.

7.6.1.6 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

Figure 53 illustrates the call flow for the case where the DND feature is deactivated.

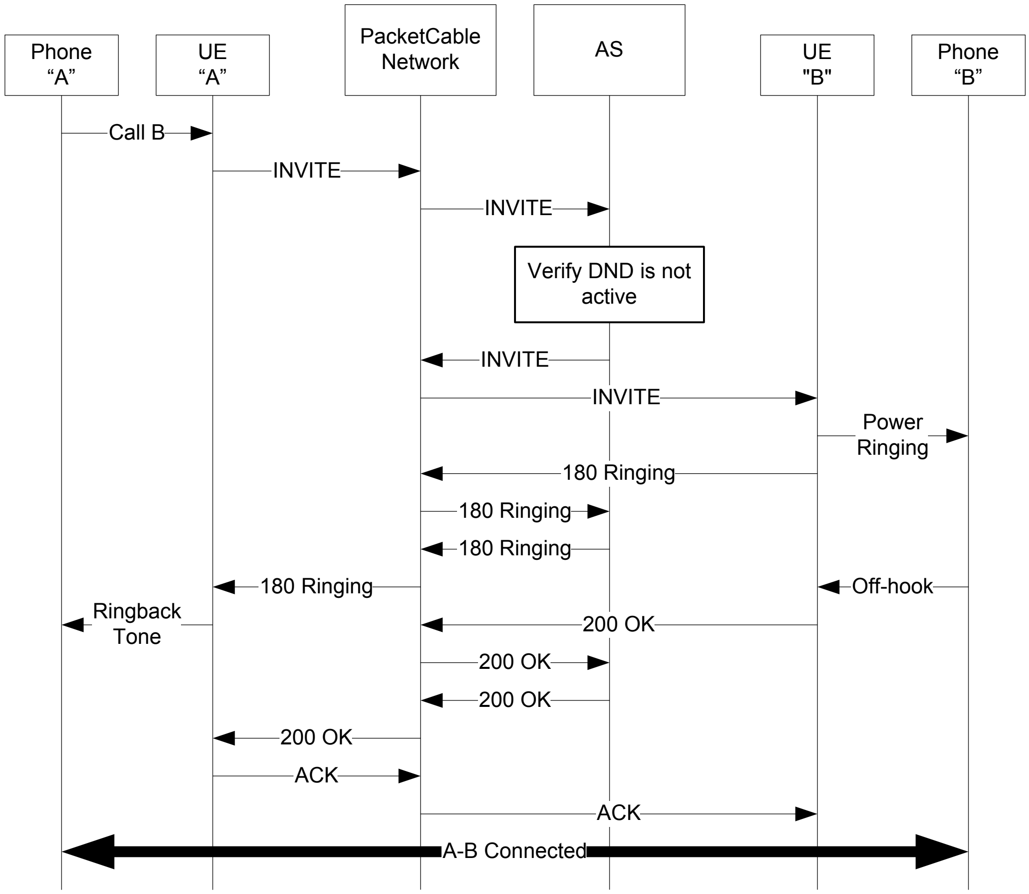

Figure 53 - Do Not Disturb (DND); Deactivated

Figure 54 illustrates the call flow for the case where the DND feature is activated and the caller either has the DND override privilege or is on the DND exempt list.

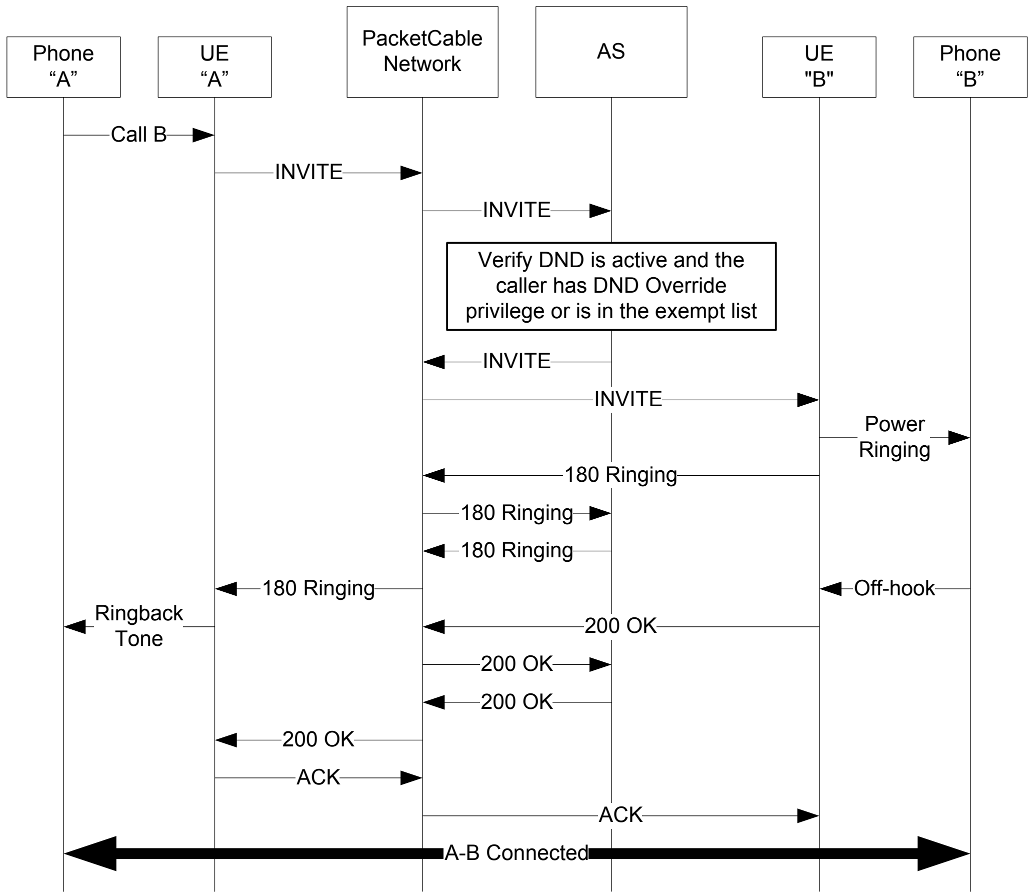

Figure 54 - DND - Override/Exempt Privilege

Figure 55 illustrates the call flow for the case where the DND feature is activated and the caller does not have a valid override PIN.

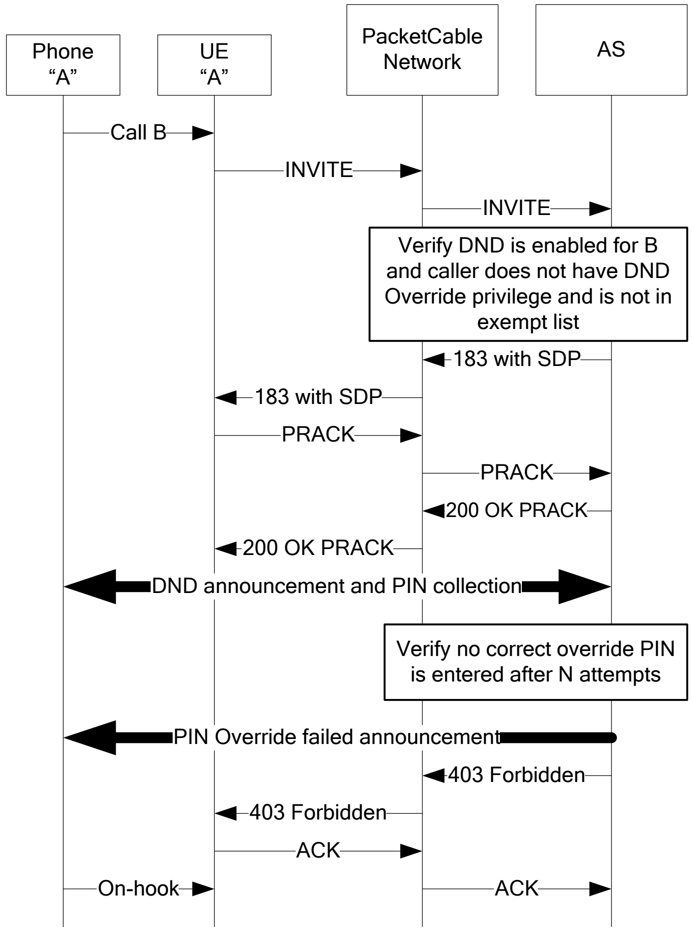

Figure 55 - DND; Invalid Override PIN

Figure 55 illustrates the early media between the caller and the AS. In actuality, the media server handling the DND announcement may not be integrated with the AS. Rather, it may be a standalone media resource device that is utilized by the AS; in this case, the call flow can deviate from what is shown in Figure 55.

Figure 56 illustrates the call flow for the case where the DND feature is activated and the caller has a valid override PIN.

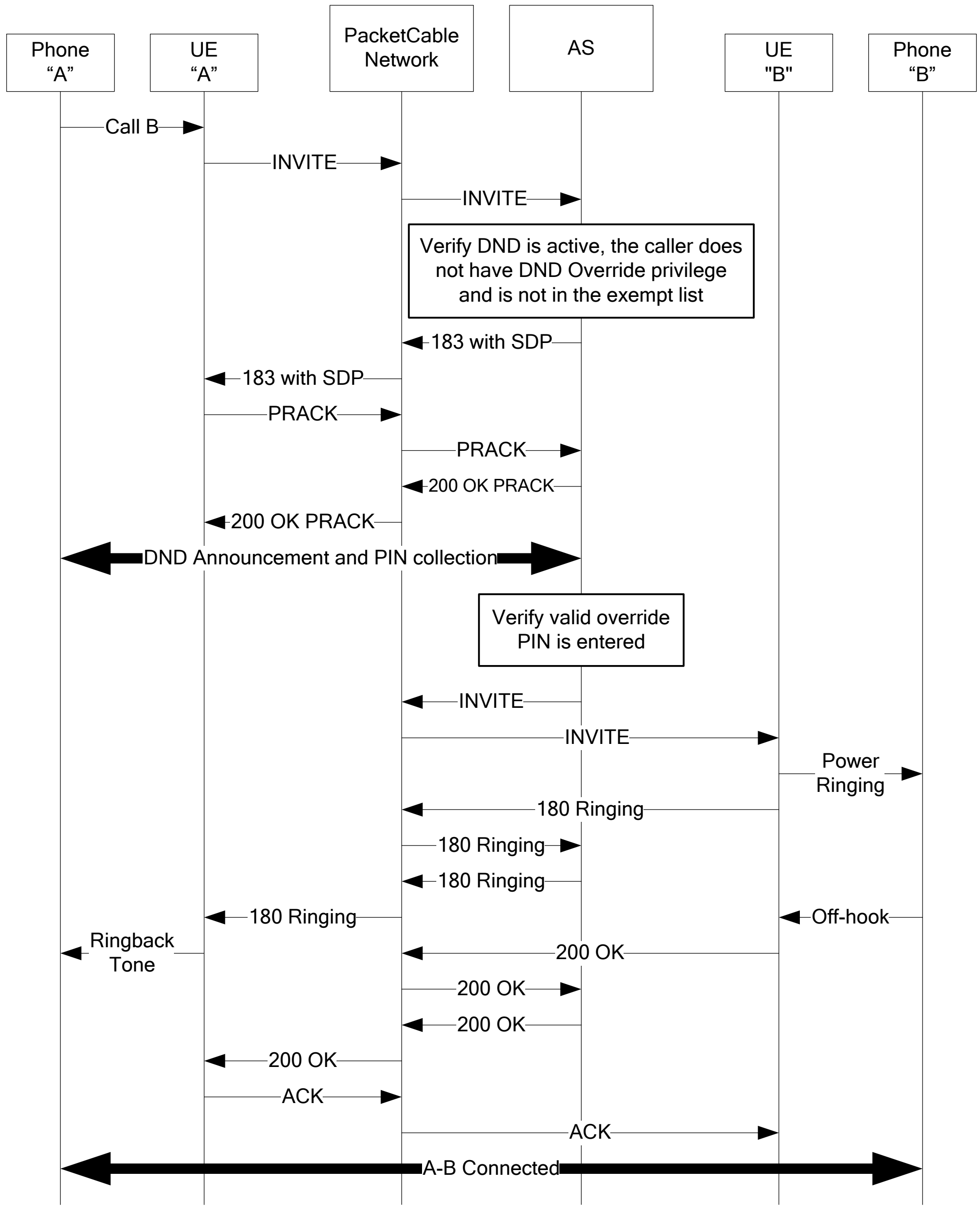

Figure 56 - DND - Valid Override PIN

7.6.2 Subscriber Programmable PIN

7.6.2.1 Feature Description

The subscriber programmable Personal Identification Number (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 Subscriber Programmable PIN (SPP) feature provides an interface for the subscriber to change PIN codes for different features on an as needed basis (that is, 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 with a different P-Asserted-Identity from changing the subscriber's PIN.

If the subscriber is attempting 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, the subscriber interacts with the service provider's SPP interface. This interface can be a web based or an interactive voice response system. Only the IVR interface is described in this specification.

7.6.2.2 Feature Activation/Deactivation

The SPP feature is not provisioned by itself, but is made available whenever a feature requiring SPP is made available in the UE. Examples of features requiring SPP are Do Not Disturb and Outbound Call Blocking. Whenever a feature requiring SPP is provisioned on the UE, the procedures for handling the SPP VSC should also be provisioned on the UE (this may include any combination of digit maps combined with locally-stored procedures).

7.6.2.3 Feature Execution

The user can access the SPP feature via the telephony user interface (that is, the phone keypad) by entering a VSC. The SPP VSC is defined for the UE as part of the Digit Map defined in Section 7.1.1. Upon matching the user-entered digits with the SPP VSC, the UE carries out the SPP-PROGRAM action. If the feature availability status indicates that the SPP feature is not available, then the UE MUST NOT send the VSC to the SPP AS. If the feature availability status indicates that the SPP feature is available, then the UE MUST send the VSC to the SPP AS via the mechanism specified in Section 7.1.2. The SPP AS MAY, in turn, connect the user to an IVR system for SPP programming.

Upon receipt of an INVITE identifying the SPP feature, and if the SPP feature is determined to be active for that user, the SPP AS MUST send back to the caller a reliable Session Progress (183) response with an early session disposition type as defined in [RFC 3959], and appropriate SDP content for a 2-way early media. At this point, the exact interaction between the SPP AS and the user is left to the service provider.

Upon completion of the SPP process, the SPP AS MUST terminate the session by sending a 487 Request Terminated message.

7.6.2.4 Feature Data

Table 36 summarizes the feature data defined to support implementation of the SPP feature.

Table 36 - Subscriber Programmable PIN Feature Data

Data	Type	Persistence	Scope	Stored By	Written By	Read By	PACM Requirement
Feature Availability Status	Boolean	Volatile on UE, Non-volatile on XDS	Per public user ID	XDS	XDS	UE	Mandatory

7.6.2.5 Feature Interactions

The following feature interaction applies to SPP:

- Once SPP programming is started, flash processing by the UE MUST be disabled.

7.6.2.6 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

Figure 57 provides an informative call flow of the Subscriber Programmable PIN editing feature. This flow assumes a voice- and DTMF-capable IVR with the ability to understand audible and DTMF responses. PIN editing exchanges are usually performed using such exchanges.

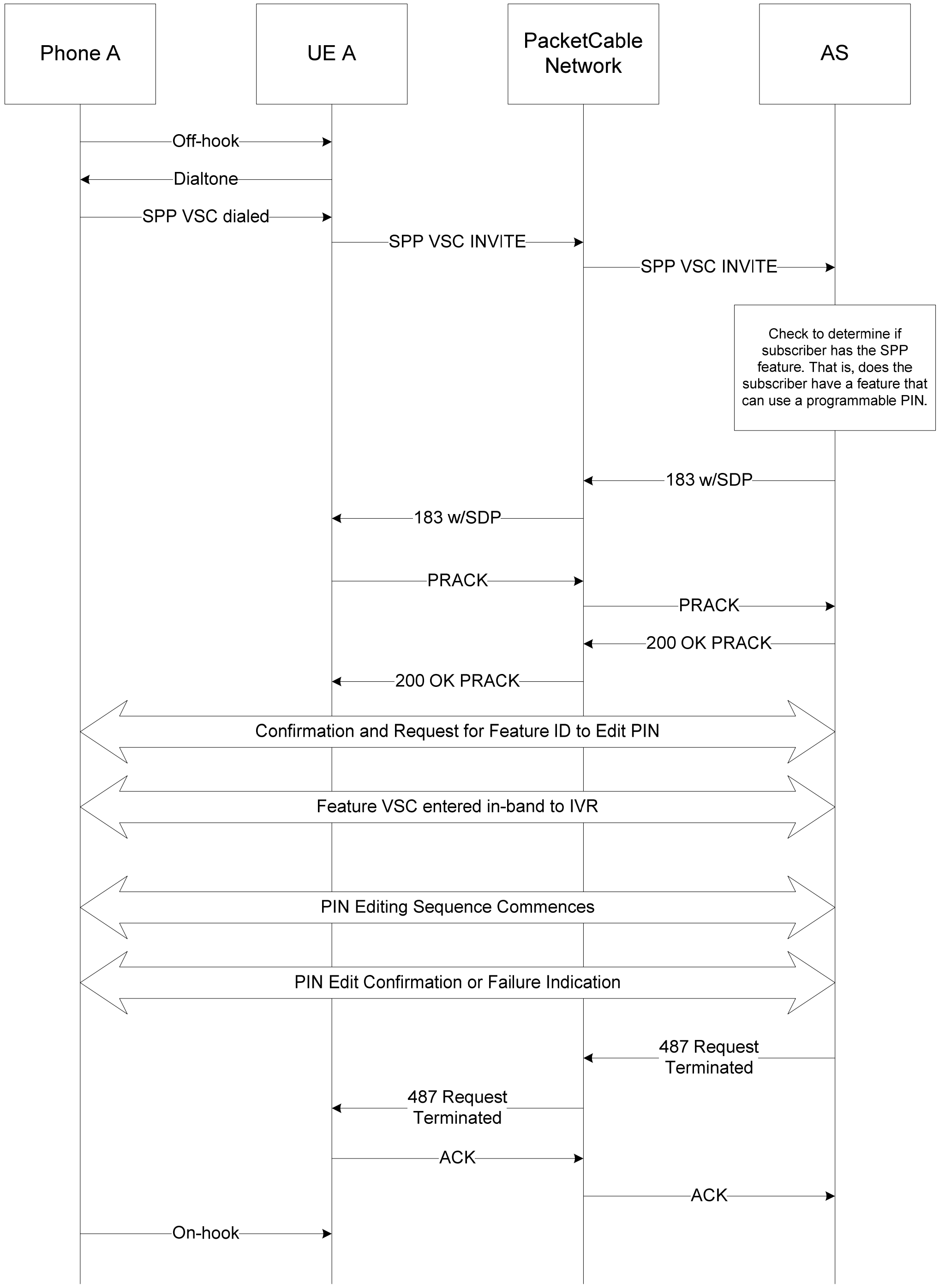

Figure 57 - Subscriber Programmable PIN (SPP)

7.6.3 Distinctive Alerting

7.6.3.1 Feature Description

The Distinctive Alerting (DA) feature allows different alerting patterns to be rendered to the user on a per call basis. The selection of a particular alerting pattern for a given incoming call is determined by the Distinctive Alerting

Application Server (DA AS) in the terminating network. The overall execution of the feature is carried out jointly between the DA AS and the called UE.

The alert-selection logic on the AS may be provisioned by the service provider, a third-party service provider, or by the users themselves. For instance, a user may specify such logic via a web portal, and may select the alerting pattern based on the caller information.

The DA AS inserts a reference to the selected alerting pattern in the incoming INVITE and forwards the INVITE to the called UE (via the terminating S-CSCF), which in turn renders the specified alerting to the user.

Distinctive Alerting is dependent upon the Screening List Editing (SLE) feature. A subscriber can initiate procedures for activating, deactivating, modifying or obtaining a status report for Distinctive Alerting by going off-hook, receiving a dial-tone, and dialing the Distinctive Alerting VSC (typically *61 for activation and *81 for deactivation). In the past, DA had separate activation and deactivation VSCs, but Telcordia standards have changed to support a single VSC that provides access to the SLE feature for updating the DA feature's configuration. In order to maintain backward compatibility with the older, two-VSC method for DA activation/deactivation, two VSCs may still be used but either VSC will invoke the same SLE feature for updating the DA feature's configuration. This allows operators who currently have customers using *61 and *81 as separate activation and deactivation codes to continue such use while permitting other operators who are not currently providing screening list features to advertise just one code as a single feature code.

[GR 219] describes the SLE feature for Distinctive Ringing/Call Waiting.

7.6.3.2 Feature Activation and Deactivation

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

The dynamic activation and deactivation of the Distinctive Alerting feature is not required.

With the Distinctive Alerting feature activated, the user may be able to configure the alert patterns and their selection criteria via a web portal.

With the Distinctive Alerting feature activated, the UE MUST be able to initiate an SLE session to configure the public identities that cause distinctive alerts when an INVITE from any of these public identities is received. Once either Distinctive Alerting feature code has been successfully entered as part of the DA-MAINT digit map action, the UE checks to see if the Distinctive Alerting feature is available. If the feature is available, then the UE MUST send an INVITE with the request URI containing the VSC as per the network requirements. If the feature is unavailable, the UE MUST NOT send an INVITE.

The subscriber receives announcements providing the following information (not necessarily in this order):

- The name of the service (that is, Distinctive Alerting)
- The current size of the subscriber's Distinctive Alerting list
- Actions and associated dialing codes available to the user:
    - Add entr(y)ies to the list
    - Delete entr(y)ies from the list
    - List review

7.6.3.3 Feature Execution

The feature execution of the Distinctive Alerting is carried out jointly by both the terminating Distinctive Alerting AS and the called UE, with the former performing the selection of an alert pattern for the incoming call and the latter appropriately rendering the pattern to the user.

For an incoming INVITE request, the DA AS checks to see if the subscriber identified by the request URI has the DA feature activated. If the feature is not activated for the subscriber, the DA AS MUST NOT perform any Distinctive Alerting processing before forwarding the INVITE to the called party.

If the feature is activated for the subscriber, and the call is determined to receive a distinctive alert, the DA AS checks the caller's Caller ID display status against the distinctive alerting precedence over Caller ID blocking. If the distinctive alerting precedence over Caller ID blocking is enabled, the DA AS MUST insert an Alert-Info header into the INVITE request to convey the selected alert that the called UE needs to apply. If the distinctive alerting precedence over Caller ID blocking is disabled and the caller has indicated Caller ID blocking, the DA AS MUST insert an Alert-Info header into the INVITE request indicating the default alert pattern. If the distinctive alerting precedence over Caller ID blocking is disabled and the caller has not requested Caller ID blocking, the DA AS MUST insert an Alert-Info header into the INVITE request to convey the selected alert which the called UE needs to apply.

Upon receiving the incoming INVITE request with an Alert-Info header parameter, the called UE MUST alert the user according to the pattern carried in the Alert-Info header parameter. If the Alert-Info header is not present in the request, the UE SHOULD apply a default alert pattern to the user.

The UE MUST map the value received in the Alert-Info header into a well-defined signal (either audio or visual) and render it to the user. If the UE determines that the value of the Alert-Info header is in error, the UE MUST play a default alert pattern instead.

A baseline set of alert patterns are defined in Appendix A (Event Packages) of [NCS].

7.6.3.4 Feature Data

Table 37 summarizes the feature data items that are defined for the implementation of Distinctive Alerting.

Table 37 - Distinctive Alerting Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Feature Availability Status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
Distinctive Alerting precedence over Caller ID blocking	Boolean (0=Caller ID blocking has precedence, 1=Distinctive Alerting has precedence)	Non-volatile	Per public user ID	AS	AS	AS	None

7.6.3.5 Feature Interactions

Call Waiting

When the Call Waiting feature is activated for the called UE and the Distinctive Alerting AS has access to the Call Waiting activation status, the Distinctive Alerting AS MAY select a special alert that can either provide distinctive ringing or a distinctive call waiting alert to the called party. This allows the UE to provide the appropriate alert, depending on whether the called party is engaged in an active call or not. The Distinctive Alerting AS provides this dual-mode alert by populating the Alert-Info header with a well-known value that the UE can interpret based upon whether it is in an active call. The called UE MUST map the dual-mode alert specified by the Alert-Info header to a distinctive ringing alert if the UE is idle. The called UE MUST map the dual-mode alert specified by the Alert-Info header to a distinctive call waiting tone if the UE is engaged in an active call.

Caller ID Display Blocking

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

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

NOTE: Telcordia [GR 219] specifies that Distinctive Alerting should take precedence over Caller ID Display Blocking, in that the called party receives the distinctive alerting treatment but does not have the number displayed. This specification also allows the reverse precedence to occur if desired, so that Distinctive Alerting is not provided when the caller requests anonymity.

Besides the alert selection for the incoming call and the other interactions specified in this section, the Distinctive Alerting feature does not impact the execution of any other features. In particular, even with the presence of the Alert-Info header in the INVITE request, the UE MUST NOT render the alert to the user if the normal execution of other activated features require that the user not be alerted.

On the other hand, if the normal execution of other activated features requires that the user be alerted, the Alert-Info header MUST be honored in rendering the alert to the user.

7.6.3.6 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

Figure 58 illustrates a call flow for Distinctive ringing execution.

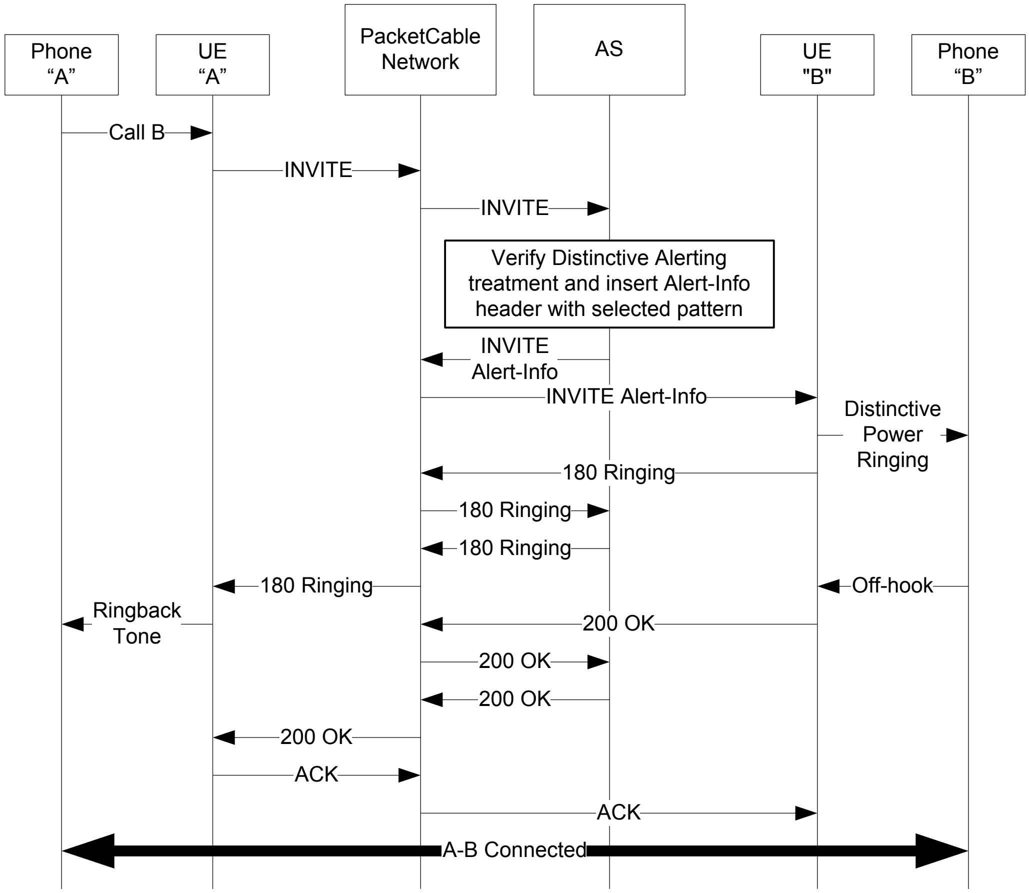

Figure 58 - Distinctive Alerting (DA); Distinctive ringing

Figure 59 illustrates a call flow for Distinctive call waiting execution.

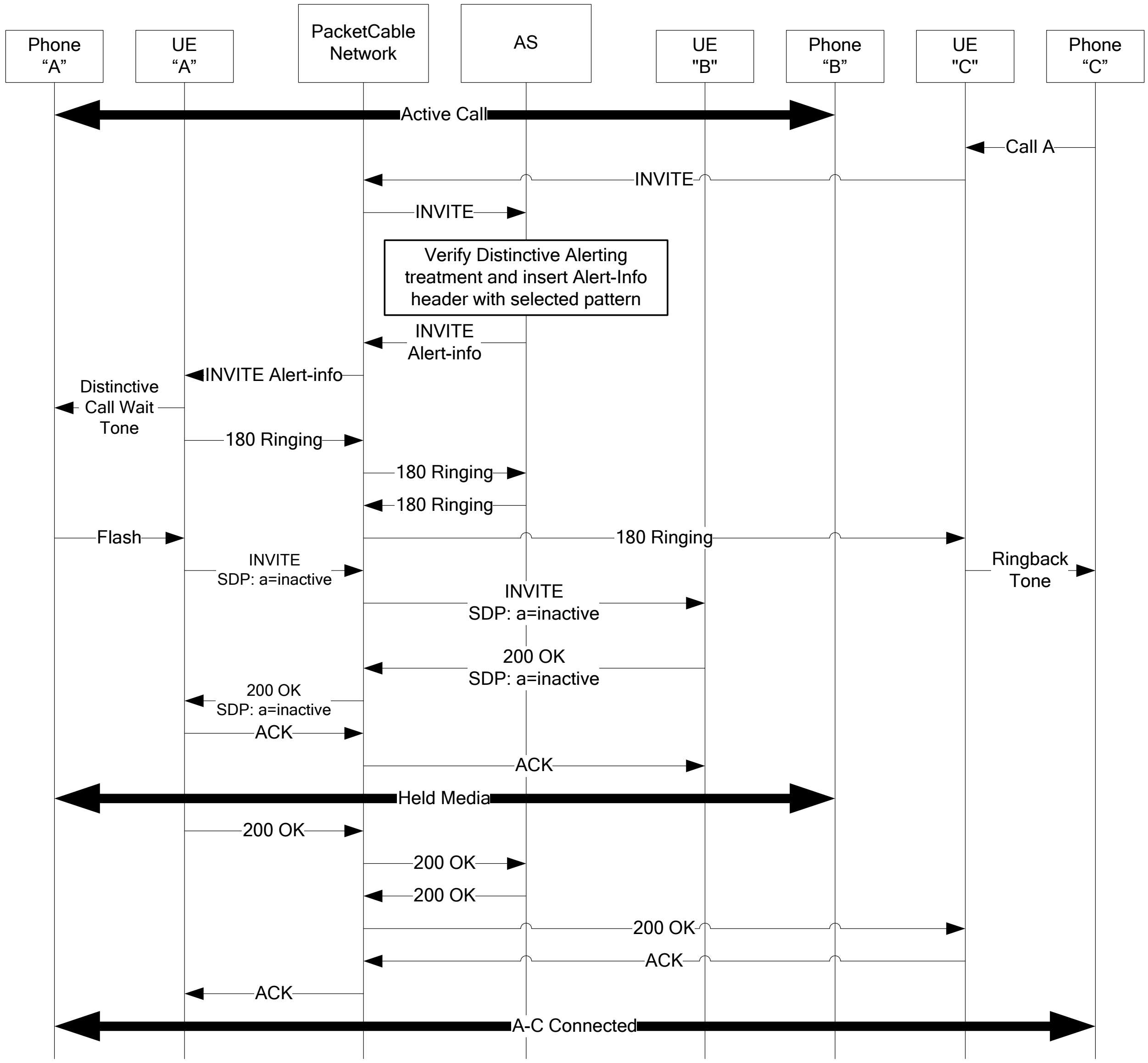

Figure 59 - DA; Distinctive call waiting

7.6.4 Message Waiting Indicators

7.6.4.1 Feature Description

Message Waiting Indicators (MWI) are used by the UE to alert the user that the status of a message account associated with the subscriber is changed. The message account is assigned to the UE by the service provider and is identified by the account's public user identity. In general, the message account can retain multimedia messages (for example, voice, fax, video, text, etc.). However, for convenience of description and without loss of generality, this specification assumes that the message account retains voice messages only; in this case, the message account can be considered as a voice-mail box.

The MWI Application Server (MWI AS) detects status changes of the message account. The actual detection mechanism is outside of the scope for this specification, but may involve the MWI AS's signaling with an embedded or standalone message storage and processing system.

The MWI AS notifies the status change of the message account to the UE as an MWI. This notification follows the general process of SIP event subscription/notification. Once received, the MWI is presented by the UE to the user as a visual signal and/or an audible signal.

7.6.4.2 Feature Activation and Deactivation

The service provider performs MWI activation and deactivation as part of service provisioning for a user. The associated provisioning procedures are outside of the scope for this specification. However, the feature activation/deactivation status is assumed to be provisioned at the MWI AS by the service provider. User-initiated activation and deactivation of the MWI feature is not required.

7.6.4.3 Feature Execution

The MWI feature is built on the basis of the SIP event subscription/notification framework. The UE and the MWI AS, as the subscriber user agent and the notifier user agent respectively, MUST meet the general PacketCable requirements for SIP event subscription/notification, as described in [PKT 24.229].

If the feature availability status is enabled, the UE MUST send out a SUBSCRIBE request, subscribing to the Message Summary and Message Waiting Indication Event Package (defined in [RFC 3842]) associated with the subscriber's message account. Once successfully subscribed to the MWI Event Package, the UE MUST refresh the subscription during the runtime before the currently active subscription is timed out. The subscription duration is configured as the Subscription Duration parameter as described in Table 38. The UE MUST use the value of the Subscription Duration, if provisioned, in the Expires header of the SUBSCRIBE request to indicate the desired length of the subscription.

The recipient MWI AS is responsible for the authorization of all incoming SUBSCRIBE requests. The complete criteria for such authorizations are specific to the service provider and are outside of the scope for this specification. However, if the requestor is not provisioned with the MWI feature, the MWI AS MUST reject the request by sending back to the requesting UE a 403 (Forbidden) or 603 (Declined) response. If the authorization is successful, the MWI AS MUST respond with a 200 OK response and start to detect the status change for the message account. As required by [RFC 3265], the MWI AS MUST also send a NOTIFY message to the UE, notifying it of the current status of the message account. Upon receipt of a SUBSCRIBE without the Expires header, the MWI AS MUST use the provisioned Subscription Duration value to populate the Expires header in the 200 OK response to indicate the subscription duration.

The MWI AS detects the occurrences of the subscribed events that are associated with the subscriber's message account. Such events are triggered to the MWI AS by the embedded or standalone message storage and processing system. If the MWI AS detects an MWI event and the subscription from the UE has not expired, the MWI AS MUST send a NOTIFY message to inform the UE of the status change of the message account. If the subscription from the UE has expired, the MWI AS SHOULD NOT send a NOTIFY message to the UE.

The formatting of the above-mentioned SUBSCRIBE and NOTIFY messages MUST conform to [RFC 3842].

When the NOTIFY message is received from the MWI AS, the UE MUST render the corresponding MWI to the user or user's CPE device, per client capabilities. The UE presentation to the user or user's CPE device could be an electrical signal (for example, telephony RJ-11 ports) and/or a visual signal and/or an acoustic (audible) signal as supported by the UE.

7.6.4.4 Feature Data

Table 38 summarizes the feature data items that are defined for the implementation of MWI.

Table 38 - MWI Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Feature Availability Status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
Subscription Duration	Integer	Non-volatile	Per network	XDS	XDS	AS, UE	Mandatory

7.6.4.5 Feature Interactions

Do-Not-Disturb (DND)

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

The MWI feature does not interact with any feature other than DND.

7.6.4.6 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

Figure 60 shows a call flow for the UE's successful subscription to the Message Summary Event Packet.

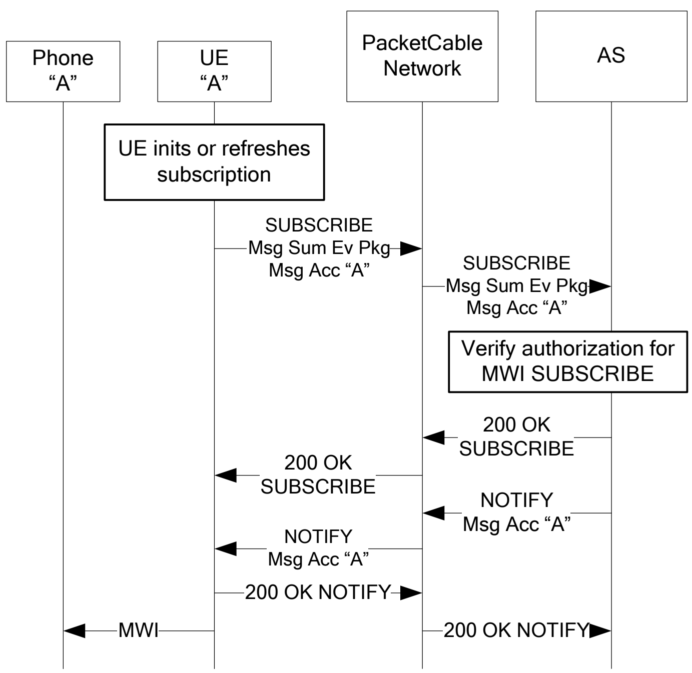

Figure 60 - Message Waiting Indicator (MWI); Successful subscription authorization

Figure 61 shows a call flow for the UE's rejected subscription to the Message Summary Event Package. The rejection was a result of the failed authorization for the MWI feature.

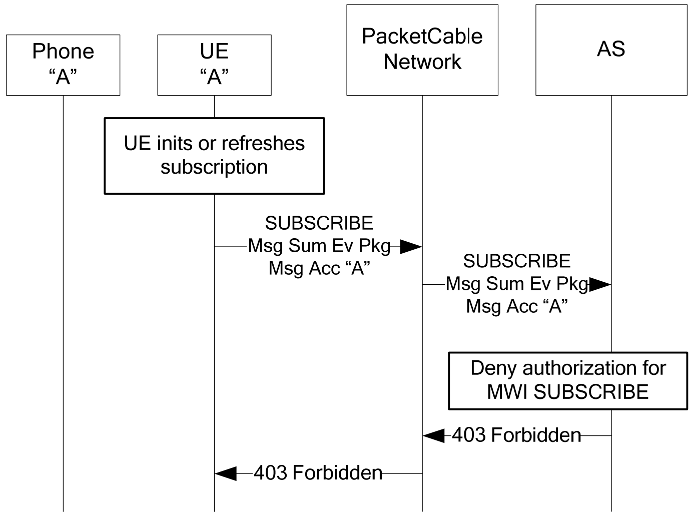

Figure 61 - MWI; Failed subscription authorization

Figure 62 shows a call flow for the MWI triggering that is a result of a new voice mail left by a caller.

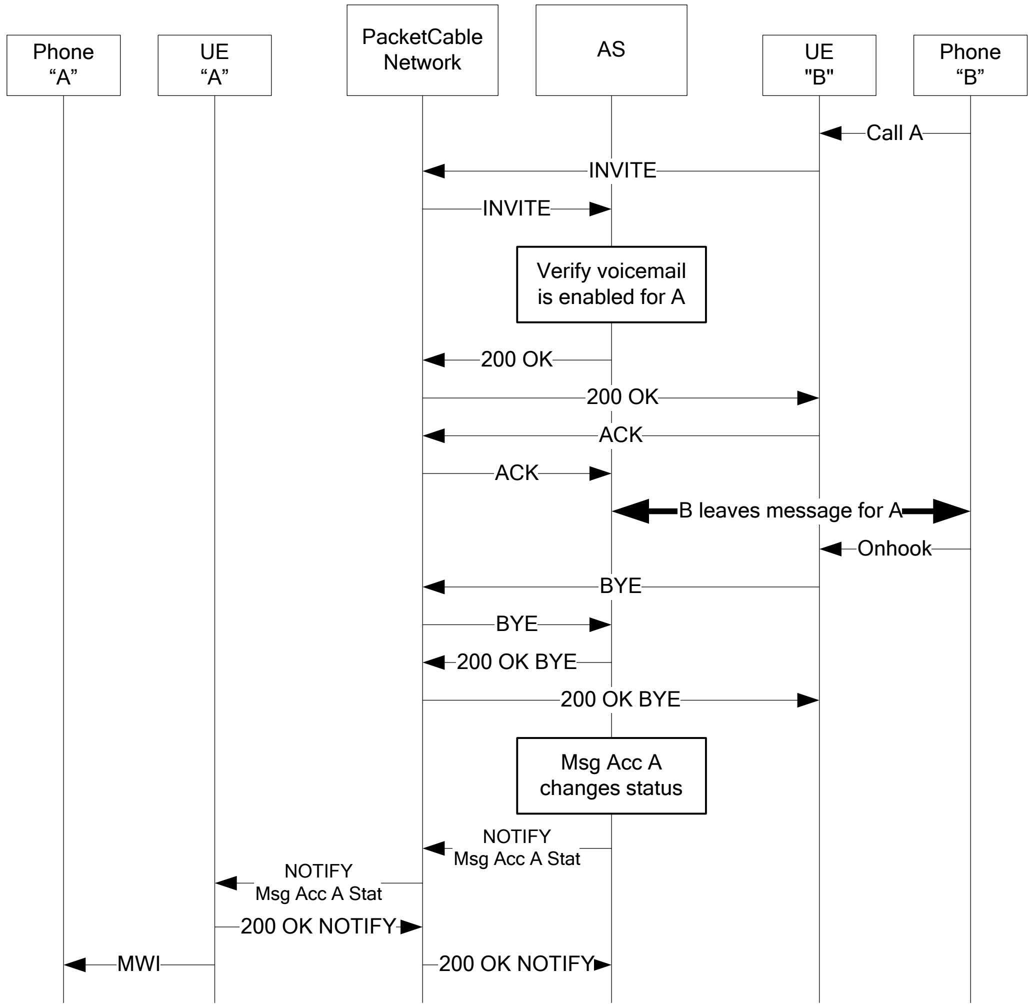

Figure 62 - MWI; Message account status change

7.6.5 Speed Dialing

7.6.5.1 Feature Description

The network based Speed Dialing feature allows the user to dial frequently-called public user IDs by assigning them either one-digit or two-digit speed-dialing codes. One-digit Speed Dialing usually accommodates eight IDs (2 - 9). Two-digit Speed Dialing can accommodate up to 99 IDs, but is usually limited to a more reasonable set of frequently-dialed IDs (for example, 30). If both one- and two-digit dialing are allowed simultaneously, then two different Vertical Service Codes (VSC) are needed (for example, *74 for one-digit Speed Dialing and *75 for two-digit Speed Dialing) for the programming of the speed dialing list.

After the speed-dialing code is entered, and a 4-second timer expires or a pound sign (#) is dialed by the subscriber, the call is placed for the user as if the ID represented by the speed-dialing code was dialed.

There are different schemes to program the frequently-dialed public user IDs associated with a UE. The IDs can be programmed by using a UE to interact with an IVR system managed by the service provider or via a web-based subscriber management function. Such programming allows for the creation of the list of IDs and the associated speed-dialing codes.

This specification adopts a network-based approach for implementing the Speed Dialing feature, with the UE-based approach being out of the scope. Nevertheless, these two approaches can co-exist in the actual products.

Refer to Telcordia LSSGR: Speed Calling (FSD-01-02-1101) for more detail on the PSTN equivalent feature. This is also known as [GR 570].

7.6.5.2 Feature Activation and Deactivation

The activation and deactivation of the Speed Dialing feature is performed by the service provider as part of the service provisioning for a user. The associated provisioning procedures are outside of the scope for this specification. However, the feature activation/deactivation status is assumed to be available to the Speed Dialing Application Server (SD AS) and the UE.

The dynamic activation and deactivation of the speed dialing feature is out of scope.

With the feature activated, the subscriber MUST be able to initiate the programming of the speed-dialing list by entering a VSC (for example, *74 for one-digit Speed Dialing and *75 for two-digit Speed Dialing). Upon matching the user-entered digits with a Speed Dialing VSC, the UE carries out the SD-PROGRAM action. The UE MUST follow the general requirements provided in Section 7.1.2 to send the VSC to the SD AS, which in turn connects the user to an IVR for the programming. Alternatively, a web portal can also be provided to the user by the service provider for programming the speed-dialing list. In either case, the requirements for the valid speed-dialing codes in Telcordia [GR 570] MUST be followed.

7.6.5.3 Feature Execution

When the user invokes the Speed Dialing feature, the UE MUST construct and initiate an appropriate INVITE with the captured speed-dialing code as the Request URI, following the requirements provided in Section 7.1.2.

Upon receiving the INVITE request, the SD AS performs the necessary authorization to determine whether or not the Speed Dialing feature is activated for the originating UE. If the authorization fails, the SD AS MUST respond to the INVITE with a 403 Forbidden response to reject the call attempt. Upon receiving the 403 Forbidden response, the UE MUST apply the reorder tone to the user.

If the authorization succeeds, the SD AS MUST map the speed-dialing code in the Request URI into the public user ID of the intended destination party. If the mapping is successful, the SD AS MUST replace the original Request URI in the received INVITE with the resulting public user ID of the intended destination party, and forward the new INVITE. If the mapping fails (that is, there is no entry corresponding to the speed-dialing digit), the SD AS MUST indicate the failure to the UE. One possible mechanism used for this failure indication is the SD AS rejecting the INVITE, so that the UE plays an error tone or announcement. Another option is for the SD AS to play an appropriate announcement to the user by setting up an early media between the UE and the SD AS (or any stand-alone announcement machine).

7.6.5.4 Feature Data

Table 39 summarizes the feature data items that are defined for the implementation of Speed Dialing.

Table 39 - Speed Dialing Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Speed Dialing Map Table	Table	Non-volatile	Per public user ID	AS	AS	AS	None
Feature Availability Status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
VSC for Programming 1-Digit Speed Dialing	VSC	Non-volatile	Per public user ID	AS	AS	AS	None
VSC for Programming 2-Digits Speed Dialing	VSC	Non-volatile	Per public user ID	AS	AS	AS	None

7.6.5.5 Feature Interactions

Outbound Call Blocking

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

Three-Way Calling

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

Call Forwarding

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

7.6.5.6 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

Figure 63 shows a call flow for the normal execution of the Speed Dialing feature:

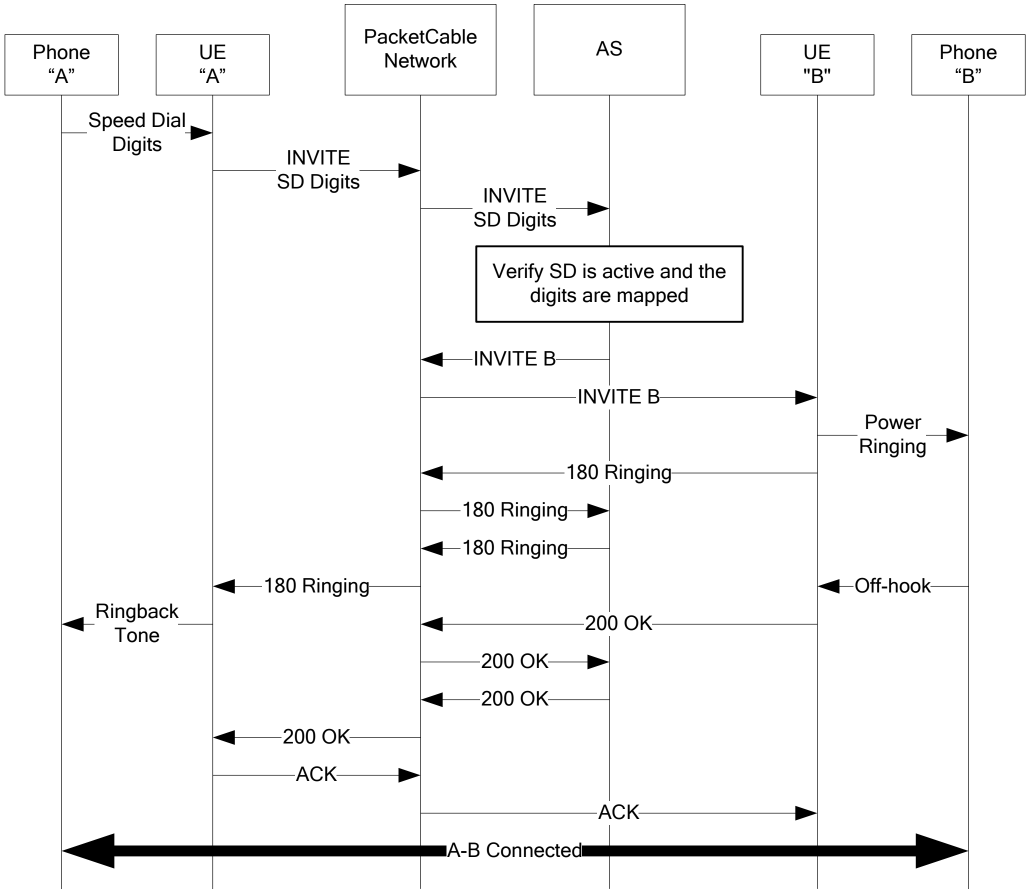

Figure 63 - Speed Dialing (SD)

For the Speed Dialing Feature, the AS does not need to record the route. Therefore, only the responses associated with the INVITE request are sent back to the AS.

Figure 64 shows a call flow for the case of failed authorization for the Speed Dialing feature.

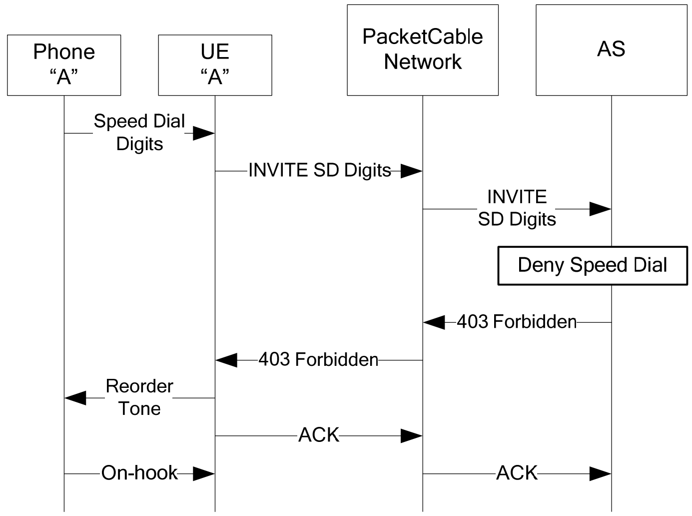

Figure 64 - SD; Failed authorization

Figure 65 shows a call flow for the case where the user-provided speed-dialing digit cannot be mapped into a destination public user ID.

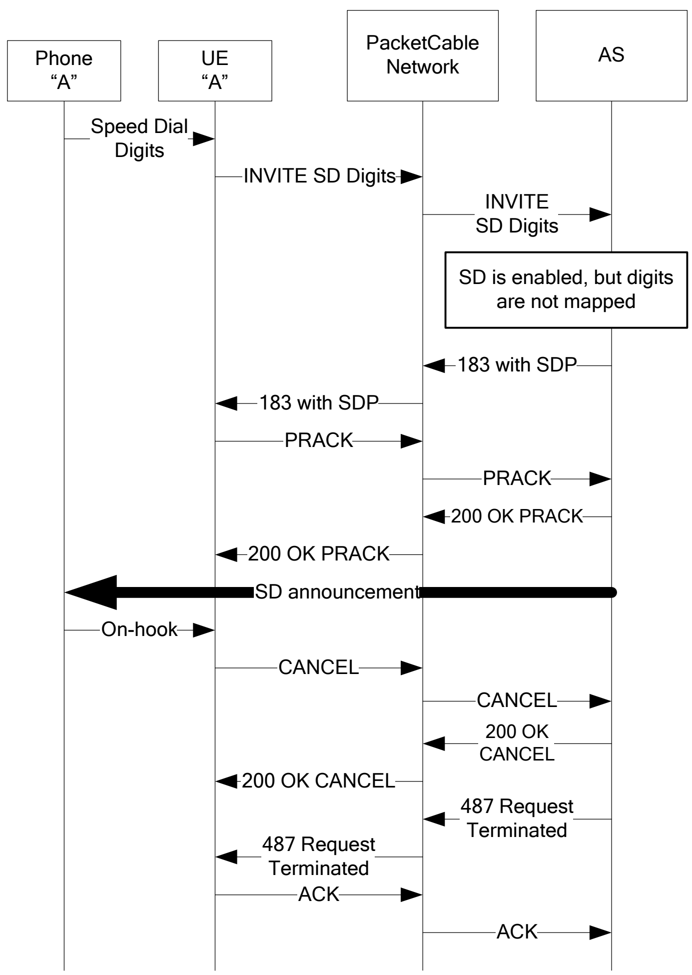

Figure 65 - SD; No Mapping

7.6.6 Customer-Originated Call Trace

7.6.6.1 Feature Description

The customer-originated call tracing (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 feature adds the capability for 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 or calls.

The intent of this COT specification is to simulate the user behavior and general feature operation as defined in [GR 216].

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) to begin the tracing activities. The COT customer needs to activate the feature immediately, prior to the arrival of a new incoming call, since only the information of the last call is stored by the COT feature.

7.6.6.2 Actions Prior To Feature Execution

Whenever the UE has been configured to provide the COT feature, via the Feature Availability Status parameter, the UE MUST store the public identity of the most recent non-emergency and non-operator caller contained within the P-Asserted-Identity header, Dialog-ID, and time stamp of the last calling party for which the call was answered. The UE MAY persistently store this information. If the most recent non-emergency and non-operator caller was anonymous (either because the caller blocked the delivery of their identity, or because the identity was unavailable), the UE MUST store the call ID, as presented in the CallID header of the anonymous call.

Further, whenever a public identity is subscribed to the COT feature, all anonymous calls placed to the public identity are sent to the COT Application Server (COT AS). This allows the COT AS to log the most recently blocked public identity (which is contained in the P-Asserted-Identity header of the INVITE). The COT AS stores call data for all terminating calls that originate from a UE who has privacy invoked and for which a 200 OK was received from the called UE. The COT AS MUST record both the most recent terminating call that originates from a UE who has privacy invoked, and any additional terminating calls that originate from a UE who has privacy invoked that arrive within the defined Call History Time Period. It is recommended that the Call History Time Period be set to at least two minutes.

7.6.6.3 Feature Execution

The user can activate the COT feature via the telephony user interface (that is, the phone keypad) by entering a VSC. The COT VSC is defined for the UE as part of the Digit Map defined in Section 7.1.1. Upon matching the user-entered digits with the COT VSC, the UE carries out the COT-ACTIVATE action. If the feature availability status indicates that the COT feature is not available, then the UE MUST NOT execute the COT feature request.

If the feature availability status indicates that the COT feature is available, then the UE checks the identity of the most recently answered caller. If the identity is known to the UE, the UE MUST send an INVITE to the COT AS using the mechanism specified in Section 7.1.2. In addition, the UE MUST provide the calling UE's public identity (P-Asserted-ID), Dialog-ID, and the time stamp of the offending caller to the COT AS via the P-DCS-TRACE-PARTY-ID header as defined in Section 7.6.6.4.

If the feature availability status indicates that the COT feature is available, then the UE checks the identity of the most recent caller. If the identity is unknown to the UE, the UE MUST send an INVITE with the request URI containing the COT activation code and the call ID of the most recent call answered by this UE. The UE MUST use the following syntax for a SIP URI that includes a VSC and a call ID:

```
sip:<VSC>.<call-id>@<domain>;user=dialstring
```

Where VSC is "*" followed by the two-digit code used to invoke COT, the call-id is the SIP call id of the most recently answered call, and the domain is the domain name of the subscriber.

The following is an example of such a SIP URI:

```
sip:*57.a84b4c76e66710%40provider.atlanta.com@mydomain.com;user=dialstring
```

The "@" sign in the call-id in the above example is escaped as "%40" so that the call-id can conform with standard URI syntax.

Upon receipt of an INVITE identifying the COT feature, the COT AS checks to see if the P-DCS-TRACE-PARTY-ID header is present. If the header is present the COT AS MUST prompt the user to confirm that the trace is wanted.

If the received INVITE does not contain the P-DCS-TRACE-PARTY-ID header, the COT AS searches its network-based call logs to discover the identity of the anonymous caller identified in the request URI. If the COT AS is not able to find the identity of the caller matching the call ID provided by the requester, the COT AS MUST fail the COT request by playing an error announcement to the requester, according to the procedures specified in Section 7.1.6. If the COT AS does find the identity of the caller matching the call ID provided by the requester, the COT AS MUST prompt the user to confirm that the trace is wanted.

Upon confirmation of the feature by the subscriber, the AS MUST complete the feature execution and play a confirmation message indicating success or failure of the trace collection procedure.

If confirmation is not provided by the subscriber, the AS MUST NOT complete the feature execution.

7.6.6.4 COT ID Header Format

The following provides the format of the P-DCS-TRACE-PARTY-ID header described in RFC 3603:

```
P-DCS-Trace-Party-ID = "P-DCS-Trace-Party-ID" HCOLON
                                  name-addr *(SEMI trace-param)
trace-param = callid-param
             / remote-param ; from draft—sip-target-dialog-03
             / local-param ; from draft—sip-target-dialog-03
             / timestamp-param
             / generic-param

callid-param = "call-id" EQUAL callid ; callid from RFC3261
timestamp-param = "timestamp" EQUAL 1*DIGIT ["." 1*DIGIT]

Header field         where proxy  ACK  BYE  CAN  INV  OPT  REG
------------         ----- -----  ---  ---  ---  ---  ---  ---
P-DCS-Trace-Party-ID   R     dr    -    -    -    o    -    -


                                        SUB  NOT  REF  INF  UPD  PRA
                                        ---  ---  ---  ---  ---  ---
                                         -    -    -    -    -    -
```

The address contained in name-addr contains the P-Asserted-ID (public identity) of the caller for the most recently answered call. Thus the addr-spec typically contains a SIP or TEL URI of the alleged malicious caller.

The dialog-ID is described in [RFC 3261] (SIP). An example of the header is:

```
P-DCS-Trace-Party-ID: call-id=a84b4c76e66710; timestamp=123456789;
                           local-tag=a6c85cf; remote-tag=1928301774
```

The COT-invoking UE MUST populate the timestamp field using the Simple Network Time Protocol (SNTP) timestamp as defined in [RFC 4330].

7.6.6.5 Feature Data

Table 40 summarizes the feature data defined to support the implementation of the COT feature.

Table 40 - Customer Originated Trace Feature Data

Data	Type	Persistence	Scope	Stored By	Written By	Read By	PACM Requirement
Feature Availability Status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
P-Asserted Identity Header	ASCII string	Non-volatile	Per session	AS or UE (Note 1)	P-CSCF, S-CSCF	AS, UE	None
Call History Time Period	Integer (seconds)	Non-Volatile	Per network	AS	AS	AS	None
Note 1: Depends on the presentation status of the call originator							

7.6.6.6 Feature Interactions

Call Forwarding

If the malicious call was forwarded enroute to the COT subscriber, the UE captures the public identity of the originating caller, not the customer who forwarded the call.

Speed Dialing

If a user programs the call trace VSC as a speed dial number, then the COT feature is not invoked when the speed dial number is dialed. This is because speed dialing is a network based service, while COT is a UE-based service, so the Speed Dial Application Server would not have the necessary information to process a call trace request when it is received.

Auto Recall

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

7.6.6.7 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

Figure 66 shows how the COT AS collects the tracing information for all terminating calls in which the call originator has elected privacy in delivering Caller ID information to the called party.

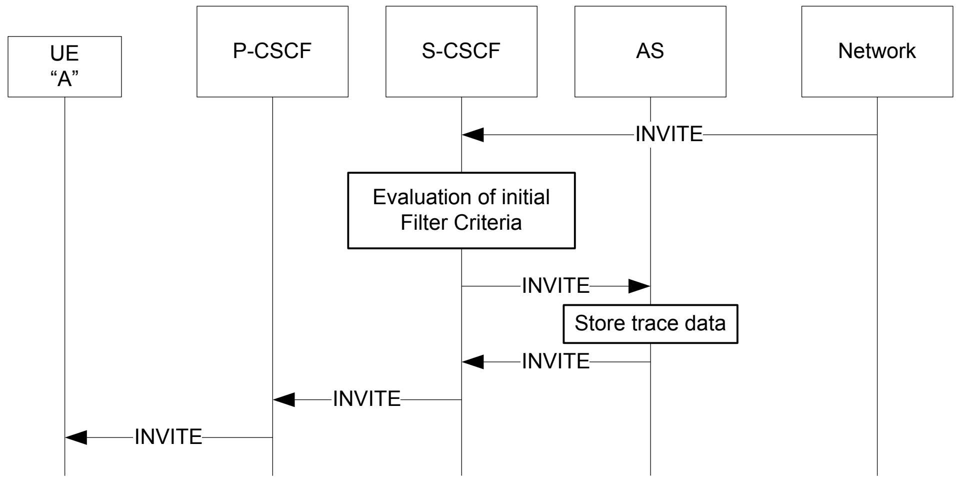

Figure 66 - COT; Collection of tracing information when caller's identity is restricted

7.6.7 Screening List Editing

7.6.7.1 Feature Description

Screening List Editing (SLE) is a set of procedures that is intended to allow customers to activate and deactivate features that use lists, obtain feature status reports, and create and modify lists of addresses in the form of Directory Number, Tel URIs or SIP URIs. Each list is associated with a particular feature to identify those calls that should be given special treatment.

UEs with analog interfaces MUST conform to the service behavior requirements in Telcordia [GR 220] unless otherwise specified here.

UEs with non-analog interfaces are allowed to provide alternative user interfaces because different types of UEs have widely varying user interface capabilities. In addition to the wide range of UEs, an operator can offer a web client mechanism to allow a subscriber to manage his or her service data. The current description is limited to interactions with the analog interface telephony UE, rather than with service management web clients.

In general, SLE supports revenue-producing features that are intended for residential as well as small business public identities. These features can support applications between networks in which signaling methods capable of transmitting calling-line identification are deployed.

SLE is used to support features such as Selective Call Forwarding (SCF – see Section 7.3.5), Distinctive Ringing/Call Waiting (DRCW – see Section 7.6.3), the Do Not Disturb white list (see Section 7.6.1), Selective Call Rejection (SCR – not specified at this time), Solicitor Blocking (see Section 7.4.3), Selective Call Acceptance (SCA – not specified at this time), and, potentially, other new feature applications as well.

Regardless of the screening-list feature that SLE is supporting, once the feature code has been processed by the AS, and the AS has connected the UE with an SLE announcement server and digit collection function, the DTMF digit interaction method used ([RFC 2833] or in-band RTP) is governed by SDP offer-answer exchange. A SLE supporting server typically operates with a voice-menu style prompt, response, and action pattern.

7.7 Call History Features

7.7.1 Auto Recall

7.7.1.1 Feature Description

The Automatic Recall (AR) feature allows a UE to automatically return a call to the last calling address (the 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. Therefore, the AR feature invocation has the following two variations:

1. **Non-Anonymous AR:** The Caller ID of the last received call at an AR requesting UE is known, and the AR can be placed at the target address directly by either entering the last Caller ID or pushing a button on the requesting UE. This case of AR feature invocation does not require the use of an AS and thus becomes of a case of basic session establishment call.
2. **Anonymous AR:** This means that the public identity of the last caller is not known to (not supplied at) the AR requesting UE, the AR feature invocation becomes that of an "Anonymous AR" feature, and involves the intervention of an AR Application Server(AR AS).

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 provides an interface to an analog telephone-like device.

7.7.1.2 Actions Prior to Feature Activation

If the AR feature is available on the UE, it MUST locally store the public identity of the most recent non-emergency and non-operator caller contained within the P-Asserted-Identity header of the INVITE and the current per-call presentation status setting, whether the call was answered or not. The UE MAY persistently store this information. If the most recent non-emergency and non-operator caller was anonymous (either because the caller blocked the delivery of the identity, or because the identity was unavailable), the UE MUST store the call ID, as presented in the CallID header, of the anonymous call.

Further, whenever a public identity is subscribed to the AR feature, all anonymous calls placed to the public identity are sent to an AR Application Server. This allows the AR AS to log the most recently blocked public identity in case the subscriber requests an AR on the call. The AR AS stores call data for all terminating calls that originate from a UE who has privacy invoked and for which a 180 ringing was received from the called UE. The AR AS MUST record both the most recent terminating call that originated from a UE who has privacy invoked, as well as any additional terminating calls that originated from a UE who has privacy invoked that arrived within the most recent X minutes, where X is a provisioned timer specified in Table 42. It is recommended that the value of X be set to at least two minutes.

7.7.1.3 Actions at Feature Activation Time

The subscriber can invoke the AR feature via the telephony user interface (that is, the phone keypad) by entering the AR VSC. The AR VSCs are defined for the UE as part of the Digit Map as defined in Section 7.1.1. Upon matching the user-entered digits with the AR VSC, the UE carries out the AR-ACTIVATE action. If the feature availability status indicates that the AR feature is not available, then the UE MUST NOT execute the AR feature request.

If the feature availability status indicates that the AR feature is available, then the UE checks the identity of the most recent caller. If the identity is known to the UE, the UE MUST send an INVITE with the request URI set to the public identity of the most recent caller, and a Call-Info header declaring purpose=answer_if_not_busy to the target identity. If the identity is unknown to the UE, the UE MUST send an INVITE with the request URI containing the AR activation code and the call ID of the most recent call attempt received by this UE. The UE MUST use the following syntax for a SIP URI that includes a VSC and a call ID:

```
sip:<VSC>.<call-id>@<domain>;user=dialstring
```

Where VSC is "*" followed by the two-digit code used to invoke COT, the call-id is the SIP call id of the most recently answered call, and the domain is the domain name of the subscriber. The following is an example of such a SIP URI:

```
sip:*69.a84b4c76e66710%40provider.atlanta.com@mydomain.com;user=dialstring
```

The "@" sign in the call-id in the above example is escaped as "%40" so that the call-id can conform with standard URI syntax.

Regardless of whether the AR is to an anonymous identity or not, the UE sets the presentation status based on the stored per-call presentation status and the procedures defined in Table 41. The resulting Presentation Status for the AR call MUST be used in accordance with the use of the privacy header as specified in Section 7.2.

Table 41 - Effects of Per-Call Presentation Features on Presentation Status for Auto Recall

PPS value for the originating UE (according to procedures in Section 7.2)	Per-call Presentation feature activated before AR Activation	Per-Call presentation feature activated before AR Reactivation	Resulting Name and Number Presentation Status
Public	None	None	Public
Anonymous	None	None	Anonymous
Public	CNDB and/or CNAB	None	Anonymous
Anonymous	CNDB and/or CNAB	None	Public
Anonymous/public	CIDS Suppression	None	Anonymous
Anonymous/public	CIDS Delivery	None	Public
Public	None/Any	CNDB and/or CNAB	Anonymous
Anonymous	None/Any	CNDB and/or CNAB	Public
Anonymous/Public	None/Any	CIDS Suppression	Anonymous
Anonymous/Public	None/Any	CIDS Delivery	Public

If the UE receives either a 180 Ringing or 200 OK in response to its INVITE, the UE MUST continue with normal call processing according to Section 7.1. If the target is anonymous, the UE MUST update its outgoing call memory with the call ID of the call, and the fact that the call is anonymous.

If the UE receives either a 486 Busy Here or 600 Busy Everywhere in response to its INVITE, the UE MUST follow the procedures described in Section 7.7.1.4.

The AR procedure re-starts for each new INVITE attempt against a new target.

For all other responses the UE MUST fail the AR request by playing an error announcement according to the procedures specified in Section 7.1.6.

Upon receipt of an INVITE identifying the AR feature, the AR AS searches its network-based call logs to discover the identity of the anonymous caller identified in the request URI. If the AR AS is not able to find the identity of the caller matching the call ID provided by the requester, the AR AS MUST fail the AR request by playing an error announcement to the requester according to the procedures specified in Section 7.1.6. If the AR AS does find the identity of the caller matching the call ID provided by the requester, the AR AS MUST replace the request-URI in the INVITE with the target identity, include a Call-Info header declaring purpose=answer_if_not_busy, add itself to the record-route header, and forward the INVITE.

7.7.1.4 SUBSCRIBE Procedures

The UE requesting AR subscribes to the target UEs dialog state using the following procedures. The UE MUST send a SUBSCRIBE to the target UE subscribing to the target's dialog event package. The UE MUST set the

SUBSCRIBE duration to a value provisioned in the UE (AR Feature timer in Table 42), and construct the request URI as described in Section 7.7.1.3.

If the target UE responds to the SUBSCRIBE with an initial NOTIFY (in addition to a 200 OK to the SUBSCRIBE), the requesting UE MUST play a delayed processing announcement (either a voice or tone announcement) according to the procedures in Section 7.1.6. If the initial NOTIFY indicates the target UE is idle, then as soon as the requesting user hangs up, the requesting UE MUST invoke the procedure in Section 7.7.1.7.

If the target UE rejects the SUBSCRIBE message from the requesting UE, then the Requesting UE MUST fail the AR request by playing an error announcement (either a voice or tone announcement) according to the procedures in Section 7.1.6.

Upon receipt of a SUBSCRIBE identifying the AR feature, the AR AS searches its network-based call logs to discover the identity of the anonymous caller identified in the request URI. If the AR AS is not able to find the identity of the caller matching the call ID provided by the requester, the AR AS MUST fail the AR request by playing an error announcement to the requester according to the procedures specified in Section 7.1.6. If the AR AS does find the identity of the caller matching the call ID provided by the requester, the AR AS MUST replace the request-URI in the SUBSCRIBE with the target identity, add itself to the record-route header, and forward the SUBSCRIBE.

If the target UE rejects the SUBSCRIBE message from the AR AS, then the AR AS MUST fail the AR SUBSCRIBE request by either returning a received response code to the requesting UE, or by playing an error announcement to the caller according to the procedures specified in Section 7.1.6.

If the target is anonymous, the NOTIFY sent as an initial response to the SUBSCRIBE is delivered to the AS that inserted the identity of the anonymous caller on the requester's behalf. When the AS receives the NOTIFY, the AS MUST replace all URIs identifying the target UE with the URI that the requesting UE used in its SUBSCRIBE message, and proxy the NOTIFY to the requesting UE.

7.7.1.5 Actions at Feature Activation Time at the Terminating AS

If a terminating AS receives an inbound INVITE with a Call-Info header declaring purpose= answer_if_not_busy, then the terminating AS MUST ignore any active CFBL service for the target public identity, not forward the call if the target is busy, and instead, deliver the INVITE to the target UE for normal AR processing. If a terminating AS receives an inbound INVITE with a Call-Info header declaring purpose= answer_if_not_busy, and if the target has CFV, SCF (where the calling party is on the SCF screening list), DND, or Solicitor Blocking (where the calling party is not on the solicitor blocking white list) active, the terminating AS MUST reject the INVITE with a 480 Temporarily Unavailable.

7.7.1.6 Actions at the AR Target UE

If the target UE receives the SUBSCRIBE while not already having the maximum number of subscriptions in progress, it MUST respond with a 200 OK response for the SUBSCRIBE and send a NOTIFY including its current busy/idle status. The target UE SHOULD NOT include dialog IDs or identities of the participating parties in the dialogs in the NOTIFY.

The target UE is provisioned with a maximum number of simultaneous SUBSCRIBEs that it will honor via the Max Simultaneous Subscriptions parameter. If the target UE receives a SUBSCRIBE message when the maximum number of subscriptions are already in progress, then the UE MUST reject the new SUBSCRIBE request with a 480 Temporarily Unavailable response code. The target UE MUST send an additional NOTIFY to the subscribing UE whenever there is a change in the dialog state of the UE, and the active dialog is ended.

7.7.1.7 NOTIFY Processing Procedures

It is often the case that, before the provisionable AR feature timer expires, the requesting UE receives a NOTIFY from the target UE indicating the target is not engaged in any dialogs. In such cases, as soon as the requesting UE is also idle, it MUST apply a special ringing as defined in [GR 227] at the local analog telephone. If the phone is answered, then the requesting UE terminates the subscription by sending a SUBSCRIBE with a value of zero in the Expires header and sends an INVITE to the target UE per the procedures in Section 7.7.1.3. Upon receiving this SUBSCRIBE, the notifier sends a 200 OK response and a NOTIFY with a Subscription-State header value of "terminated."

If the local user does not answer the phone when special ringing is applied within the AR Special Ringing Duration time, the requesting UE MUST continue its subscription to the target UE for the remaining time on the AR duration timer. If the requesting UE receives another response indicating the target UE is still idle, it MUST wait a provisionable time period (AR Special Ringing Retry Wait Interval in Table 42) and apply special ringing again. This cycle is repeated until either the requesting local user answers the phone or the AR duration timer expires.

The requesting UE MUST repeat this process until either the special ringing phone is answered, the AR timer expires, or the AR Special Ringing Retries is reached, whichever comes first, after which the AR feature is complete and the UE MUST terminate any existing subscription to the target's dialog event package by sending a SUBSCRIBE with a value of zero in the Expires header. Upon receiving this SUBSCRIBE, the notifier sends a 200 OK response and a NOTIFY with the Subscription-State header value of "terminated."

When special ringing is applied at the requesting UE, and the requesting UE has the Caller ID display capability activated, the requesting UE MUST provide the target's Caller ID if available.

In the case where the AR target is anonymous and the UE's SUBSCRIBE request is routed to an AR AS, then the AR AS is the entity receiving the NOTIFY from the target UE indicating the target is not engaged in any dialogs. When the AR AS receives this NOTIFY message, the AR AS MUST send a NOTIFY to the requesting UE, as if it were from the target originally specified by the requesting UE in its initial subscription. The NOTIFY MUST be sent with the From header set to the request URI received in the SUBSCRIBE. If there is a P-Asserted-Identity and a no privacy header with privacy=id in the NOTIFY, then the AS MUST remove the P-Asserted-Identity from the proxied NOTIFY message.

7.7.1.8 Other Scenarios and Issues

If the requesting UE receives an error response to its SUBSCRIBE message, it MUST provide an error indication to the requesting UE's user, either through a locally-stored announcement or tone, or through a streamed announcement played from a network resource, as defined in Section 7.1.6.

When a user dials the AR activation code again after the AR feature has already been activated on the local UE, if the new AR activation request is associated with the same target as the previously activated AR request, and if the previous AR request timer has not yet expired, then the UE MUST send a new SUBSCRIBE message to the target UE, with a subscription duration set to the full AR duration value provisioned in the requesting UE. This has the effect of resetting the AR request timer to the maximum feature request duration.

When a user dials the AR activation code again after the AR feature has already been activated on the local UE, if the new AR activation request is associated with a different target from the previously activated AR request, and if there are not already a number of simultaneous subscriptions outstanding equal to the Max Subscribes allowed from this public identity, then the requesting UE MUST execute the AR procedure described Section 7.7.1.3 for the new target.

If a target UE receives multiple subscriptions to its dialog event package, implying that multiple remote UEs are requesting AR or Auto Callback service targeting this UE, then there is no requirement for the UE to NOTIFY the subscribers in order of subscription (no subscription needs to have priority over another). The order in which a UE honors subscriptions is subject to implementer discretion.

7.7.1.9 Actions Required when a UE Requests Auto Recall to a PSTN Target

When the target of an AR request is on the PSTN, the requesting UE is not aware of the network that the target resides on, so the actions of the requesting UE are identical whether the target UE is in a PacketCable network or is the PSTN.

If the AR target is on the PSTN, then the initial INVITE sent by the requesting UE (in response to the requesting user dialing the AR invocation code) is forwarded to an MGC according to normal IMS routing procedures. The MGC requirements and procedures (including SIP to ISUP interworking) for support of Auto Recall are specified in [CMSS1.5].

7.7.1.10 Actions Required for a PSTN Endpoint Requesting Auto Recall to a UE Target

When a PSTN-based endpoint requests Auto Recall on a UE target, the PacketCable network receives a TCAP Network-Ring-Again request at an MGC, indicating the telephone number of the UE targeted for AR. The MGC requirements and procedures (including SIP to ISUP interworking) for support of Auto Recall are specified in [CMSS1.5].

7.7.1.11 Feature Data

Table 42 summarizes the feature data defined to support implementation of Auto Recall feature.

Table 42 - Auto Recall Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Feature Availability Status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
AR Feature Timer	Integer (seconds of feature duration, 0 to 30 minutes, default is 30 minutes)	Non-volatile on XDS, optionally Non-volatile on UE	Per public user ID	XDS	XDS	UE	Mandatory
AR Special Ringing Duration	Integer (number of special ringing ring cycles)	Non-volatile on XDS, optionally Non-volatile on UE	Per public user ID	XDS	XDS	UE	Mandatory
AR Special Ringing Retry Wait Interval	Integer (number of seconds to wait between attempts to alert the user with special ringing)	Non-volatile at XDS, optionally Non-volatile on UE	Per public user ID	XDS	XDS	UE	Mandatory
AR Special Ringing Retries	Integer (number of times to retry special ringing before canceling the AR request)	Non-volatile on XDS, optionally Non-volatile on UE	Per public user ID	XDS	XDS	UE	Mandatory
Max Simultaneous SUBSCRIBEs	Integer (max number of simultaneous subscribes the UE should send)	Volatile in UE, Non-volatile in XDS	Per public user ID	XDS	XDS	UE	Mandatory
Max Simultaneous Subscriptions	Integer (max number of simultaneous subscriptions the UE should honor)	Volatile in UE, Non-Volatile in XDS	Per public user ID	XDS	XDS	UE	Mandatory
AR Timer (X)	Integer (minutes, min 2 min)	Non-Volatile	Per public User ID	AS	AS	AS	None

7.7.1.12 Feature Interactions

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

Calling Name Delivery

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

Speed Dialing

Programming the AR activation code as a speed dial number won't invoke the AR feature correctly.

Call Forwarding Busy

An AR Request to a target with CFB activated is not allowed to be forwarded.

Do Not Disturb

An AR Request to a target that has Do Not Disturb enabled is rejected with a 480 Temporarily Failure unless the requester is on the Do Not Disturb white list.

Selective Call Forwarding

An AR Request to a target that has Selective Call Forwarding enabled is rejected with a 480 Temporarily Failure if the requester is on the Selective Call Forwarding screening list.

Call Forwarding Variable

An AR Request to a target that has Call Forwarding Variable enabled is rejected with a 480 Temporarily Failure.

7.7.1.13 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

Figure 67 illustrates an example call flow for the case where UE A targets non-anonymous target UE B for the Auto Recall feature.

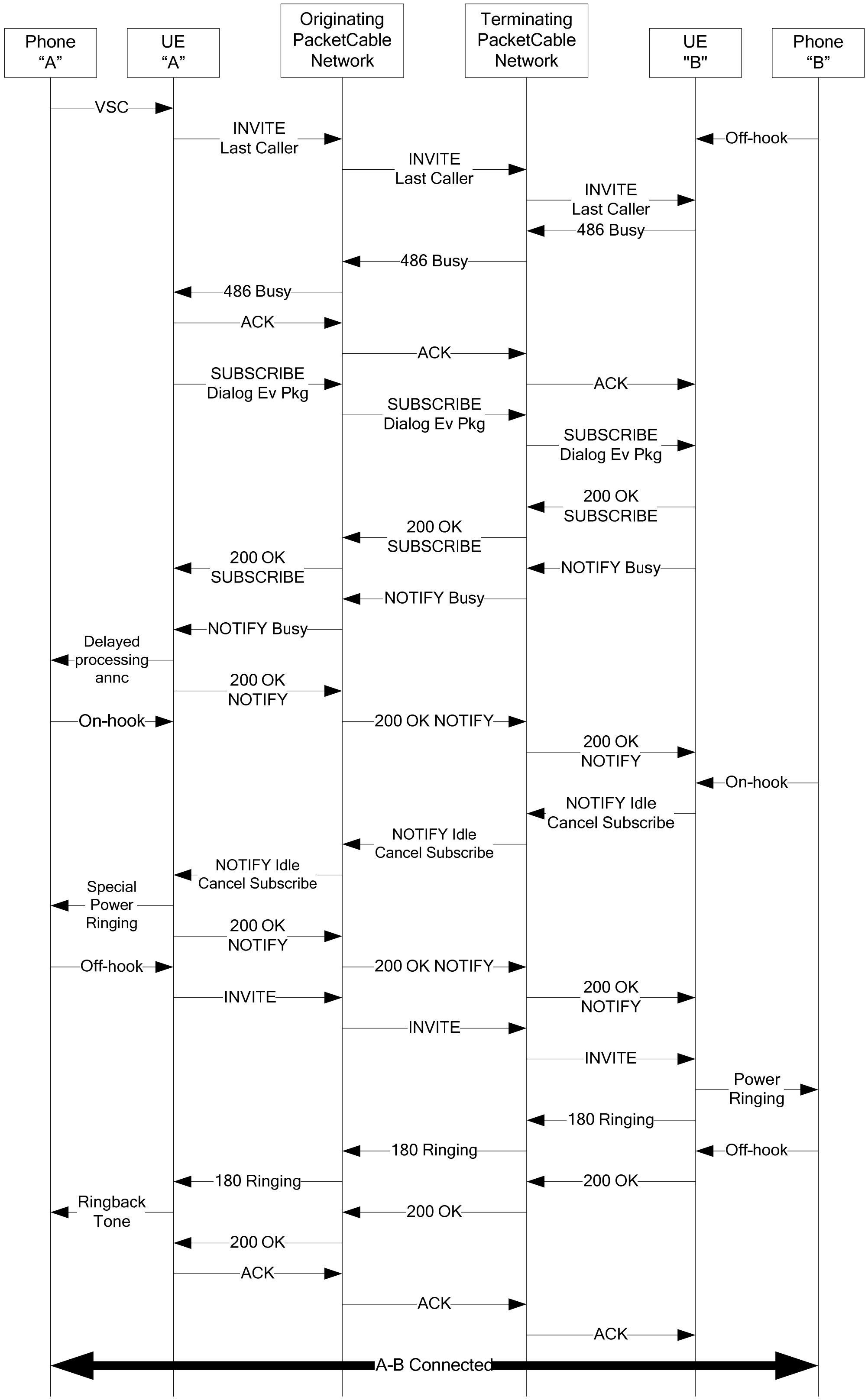

Figure 67 - Automatic Recall (AR); Target not anonymous

Figure 68 illustrates an example call flow for the case where UE A targets anonymous target UE B for the Auto Recall feature, and UE B is busy.

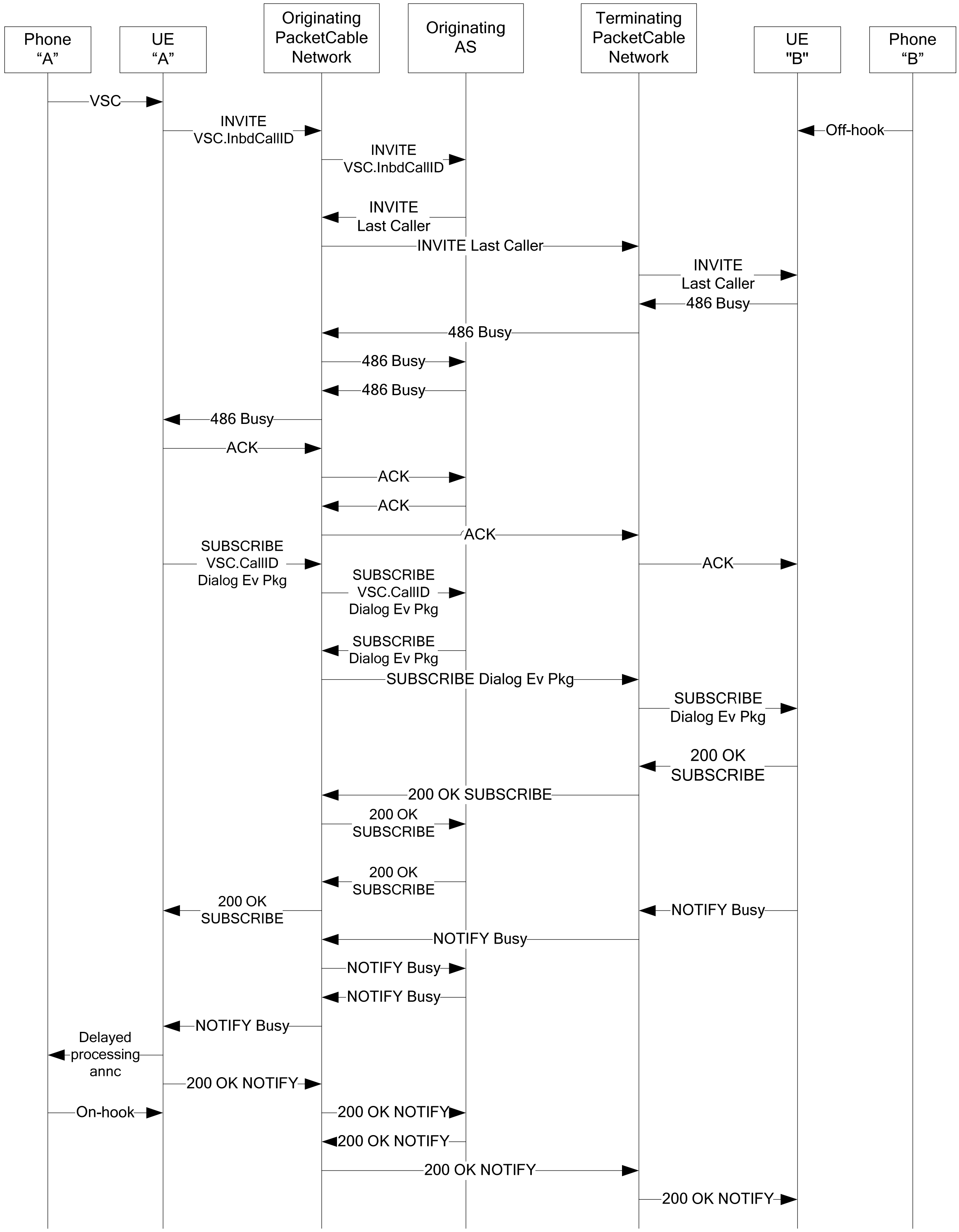

Figure 68 - AR; Anonymous target is busy

Figure 69 illustrates an example call flow for the case where the anonymous target becomes available.

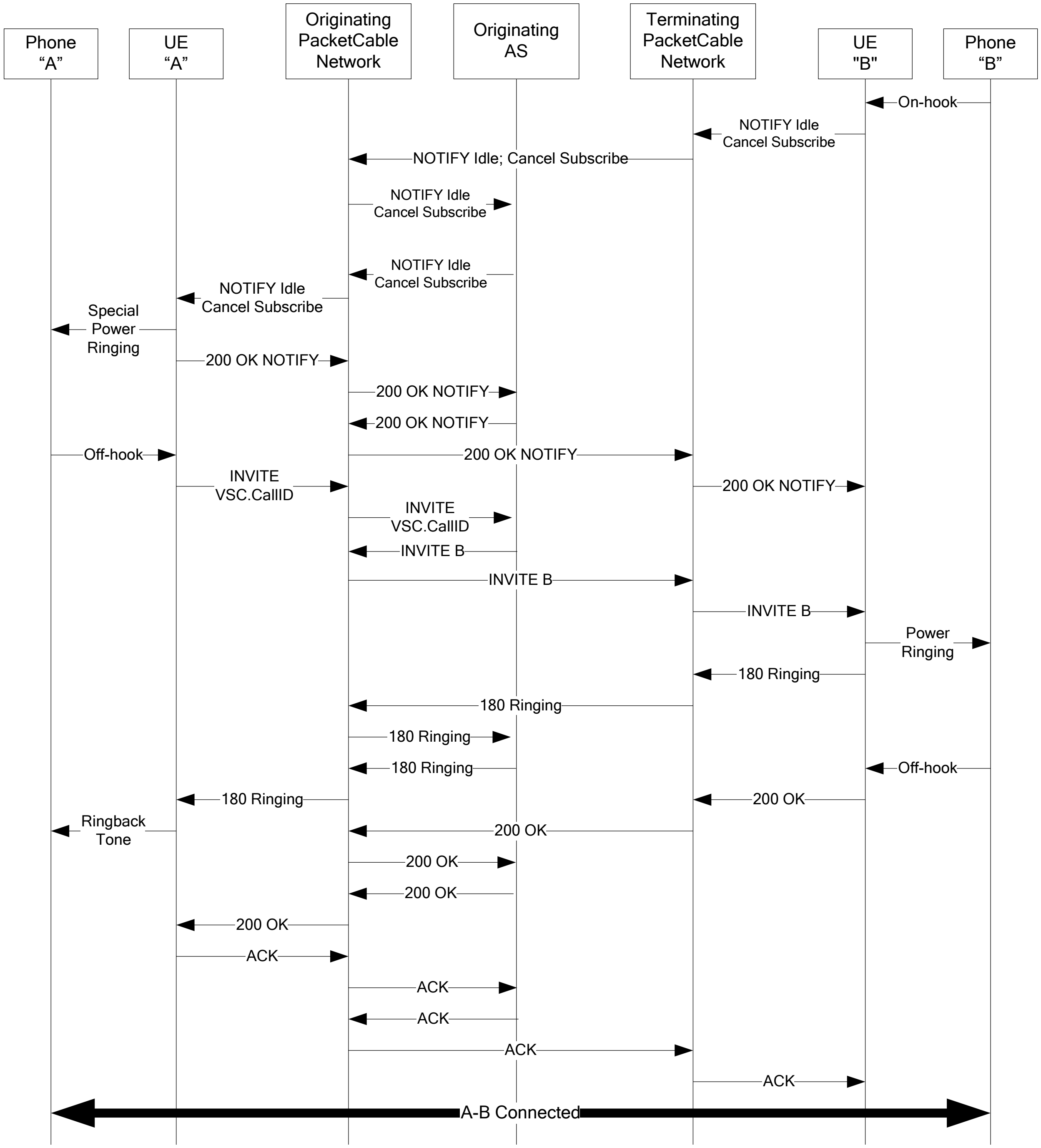

Figure 69 - AR; Anonymous target becomes idle

7.7.2 Auto Callback

7.7.2.1 Feature Description

The Automatic Callback (AC) feature allows a UE to automatically call back the last called address to which the last INVITE was sent from this UE, whether the INVITE was answered by the called party or not. The AC feature should work even when the last call made from the UE did not supply a called ID. Therefore, the AC feature invocation has the following two variations:

1. **Non-Anonymous AC:** The called ID of the last sent call from an AC requesting UE is known, and the AC call can be placed at the target address directly by either entering the last called party's public ID or pushing a button on the requesting UE. This case of AC feature invocation does not require the use of an application server (AS) and thus becomes a case of a basic session establishment call.
2. **Anonymous AC:** The called ID of the last sent call from an AC requesting UE is not known. This happens when the last call from the AC requesting UE is an anonymous Auto-Recall (AR) call. In this case, the AC feature invocation becomes that of an "Anonymous AC" feature and involves the intervention of an AC AS.

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 provides an analog interface to an analog telephone-like device.

7.7.2.2 Actions Prior to Feature Activation

Whenever the UE has been configured to provide the AC feature, the UE MUST locally store the address of the most recent non-emergency and non-operator called party, along with the caller's per-call presentation features (see Section 7.2) used at the time of the call, whether the call is answered or not. The UE MAY persistently store this information.

If the most recently called party is the target of an anonymous Auto Recall request, then the most recently called party for the AC feature is anonymous. In such a case, the UE should locally store the fact that the most recently called party was anonymous, along with the call ID of the anonymous call.

7.7.2.3 Actions at Feature Activation Time

Upon activation of the AC feature, the UE checks the identity of the most recent callee, and if that identity is known to the UE and permitted for the UE to call, the UE MUST send an INVITE with a Call-Info header declaring purpose=answer_if_not_busy to the target identity and the request URI identifying the target UE.

If the target identity is not known, the UE MUST send an INVITE with the request URI containing the AC activation code and the call ID of the most recent call attempt originated by this UE. The UE MUST use the following syntax for a SIP URI that includes a VSC and a call ID:

```
sip:<VSC>.<call-id>@<domain>;user=dialstring
```

Where VSC is "*" followed by the two-digit code used to invoke COT, the call-id is the SIP call id of the most recent answered call, and the domain is the domain name of the subscriber. The following is an example of such a SIP URI:

```
sip:*66.a84b4c76e66710%40provider.atlanta.com@mydomain.com;user=dialstring
```

The "@" sign in the call-id in the above example is escaped as "%40" so that the call-id can conform with standard URI syntax.

Regardless of whether the AC feature call is to an anonymous target UE or not, the UE sets the presentation status based on the stored per-call presentation status and the procedures defined in Table 41. The resulting Presentation Status for the AC call MUST be used in accordance with the use of the privacy header as specified in Section 7.2.

Table 43 - Effects of Per-Call Presentation Features on Presentation Status for Auto Callback

PPS value for the originating UE (according to procedures in Section 7.2)	Per-call Presentation feature activated before AC Activation	Per-Call presentation feature activated before AC Reactivation	Resulting Name and Number Presentation Status
Public	None	None	Public
Anonymous	None	None	Anonymous
Public	CNDB and/or CNAB	None	Anonymous
Anonymous	CNDB and/or CNAB	None	Public
Anonymous/public	CIDS Suppression	None	Anonymous
Anonymous/public	CIDS Delivery	None	Public
Public	None/Any	CNDB and/or CNAB	Anonymous
Anonymous	None/Any	CNDB and/or CNAB	Public
Anonymous/Public	None/Any	CIDS Suppression	Anonymous
Anonymous/Public	None/Any	CIDS Delivery	Public

If the UE receives either a 180 Ringing or a 200 OK in response to its INVITE, the UE MUST continue with normal call processing according to Section 7.1. If the target is anonymous, the UE MUST update its outgoing call memory with the call ID of the call, and the fact that the call was anonymous.

If the UE receives either a 486 Busy Here or 600 Busy Everywhere in response to its INVITE, the UE MUST follow the SUBSCRIBE procedure described in Section 7.7.1.4.

If the UE receives either a 415 Unsupported Media Type or 420 Unsupported Extension, the UE MUST follow the procedures in [RFC 3261] to attempt to complete the INVITE.

If the UE receives either a 300 Multiple Choices, 301 Moved Permanently, or 302 Moved Temporarily, the UE SHOULD retry one or more addresses according to [RFC 3261]. The AC procedure re-starts for each new INVITE attempt against a new target.

For all other responses the UE MUST fail the AC request by playing an error announcement according to the procedures specified in Section 7.1.6.

Upon receipt of an INVITE identifying the AC feature, the AC AS searches its network-based call logs to discover the identity of the anonymous last called party identified in the request URI. If the AC AS is not able to find the identity of the called party matching the call ID provided by the requesting UE, the AC AS MUST fail the AC request by playing an error announcement to the requesting UE according to the procedures specified in Section 7.1.6. If the AC AS does find the identity of the called matching the call ID provided by the requester, the AC AS MUST replace the request-URI in the INVITE with the target identity, include a Call-Info header declaring purpose=answer_if_not_busy, add itself to the record-route header, and forward the INVITE.

7.7.2.4 SUBSCRIBE Procedures

The AC AS MUST implement the requirements defined in Section 7.7.1.4.

The UE requesting AC MUST implement the requirements defined in Section 7.7.1.4.

7.7.2.5 Actions at Feature Activation Time at the Terminating AS

The AC AS MUST implement the requirements defined in Section 7.7.1.5.

7.7.2.6 Actions at the AC Target UE

The target UE MUST implement the requirements defined in Section 7.7.1.6.

7.7.2.7 NOTIFY Processing Procedure

The AC AS MUST implement the requirements defined in Section 7.7.1.7.

The UE requesting AC MUST implement the requirements defined in Section 7.7.1.7.

7.7.2.8 Other Scenarios and Issues

The UE requesting AC MUST implement the requirements defined in Section 7.7.1.8.

7.7.2.9 Actions Required when a UE Requests Auto Callback to a PSTN Target

The MGC requirements and procedures (including SIP to ISUP interworking) for support of Auto Callback are specified in [CMSS1.5].

7.7.2.10 Actions Required for a PSTN Endpoint Requesting Auto callback to a UE Target

The MGC requirements and procedures (including SIP to ISUP interworking) for support of Auto Callback are specified in [CMSS1.5].

7.7.2.11 Feature Data

Table 44 summarizes the feature data that are defined to support implementation of the Auto Callback feature.

Table 44 - Auto Callback Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Feature Availability Status	Boolean	Non-volatile	Per public user ID	AS	AS	AS	None
AC Feature Timer	Integer (seconds of feature duration, 0 to 30 minutes, default is 30 minutes)	Non-volatile on XDS, optionally Non-volatile on UE	Per public user ID	XDS	XDS	UE	Mandatory
AC Special Ringing Duration	Integer (number of special ringing ring cycles)	Non-volatile on XDS, optionally Non-volatile on UE	Per public user ID	XDS	XDS	UE	Mandatory
AC Special Ringing Retry Wait Interval	Integer (number of seconds to wait between attempts to alert the user with special ringing)	Non-volatile at XDS, optionally Non-volatile on UE	Per public user ID	XDS	XDS	UE	Mandatory
AC Special Ringing Retries	Integer (number of times to retry special ringing before canceling the AR request)	Non-volatile on XDS, optionally Non-volatile on UE	Per public user ID	XDS	XDS	UE	Mandatory

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Max Simultaneous SUBSCRIBEs	Integer (max number of simultaneous subscribes the UE should send)	Volatile in UE, Non-volatile in XDS	Per public user ID	XDS	XDS	UE	Mandatory
Max Simultaneous Subscriptions	Integer (max number of simultaneous subscriptions the UE should honor)	Volatile in UE, Non-Volatile in XDS	Per public user ID	XDS	XDS	UE	Mandatory
AC Timer (X)	Integer (minutes, min 2 min)	Non-Volatile	Per public User ID	AS	AS	AS	None

7.7.2.12 Feature Interactions

The feature interactions are specified in [GR 215].

Calling Name Delivery

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

Speed Dialing

Programming the AC activation code as a speed dial number won't invoke the AC feature correctly.

Call Forwarding Busy

An AC Request to a target with CFB activated is not allowed to be forwarded.

Do Not Disturb

An AC Request to a target that has Do Not Disturb enabled is rejected with a 480 Temporary Failure unless the requester is on the Do Not Disturb white list.

Selective Call Forwarding

An AC Request to a target that has Selective Call Forwarding enabled is rejected with a 480 Temporary Failure if the requester is on the Selective Call Forwarding screening list.

Call Forwarding Variable

An AC Request to a target that has Call Forwarding Variable enabled is rejected with a 480 Temporary Failure.

7.7.2.13 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

The AC call flows are identical to the AR call flows in Section 7.7.1.13 with the following exceptions. The public identity of the last callee (instead of the caller) is used by the initiating UE for a non-anonymous call. The call id of the last outbound call (instead of the last inbound call) is used by the initiating UE for an anonymous call.

7.8 Operator Services

PacketCable Residential SIP Telephony services require an operator services platform to be able to verify that a line is busy and to 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 provides options for interoperating with a circuit switched operator services platform.

In Busy Line Verification Service (BLV), a user calls an operator services position by dialing 0 or 0+[NPA] NXX-NXXX to determine if a line is in use. In Emergency Interrupt Service (EI), the operator provides a Busy Line Verify Service, then interrupts the 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 Service after the called party hangs up.

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

The Busy Line Verification, Emergency Interrupt, and Call Completion services MUST follow the requirements specified in [GR 1176] and [GR 531].

7.8.1 Busy Line Verification

7.8.1.1 Feature Description

From the North American network, requesters access the Busy Line Verification service 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 status of the called line. If the line is busy, the operator position bridges into the target's call in progress and listens to the called line with automatic scrambling of the called line speech 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 that case, the operator notifies the requester that busy line verification is not possible.

In the case where the line is not directly accessible by the operator services position, the line may be indirectly accessible via another, remote operator services system. In this case, the local operator services position contacts the remote operator services system and the remote operator services system performs the function. This may be implemented via a human to human telephone call between operators.

The operator must provide credentials to the network to vouch for its identity.

If the target of a BLV request has more than one UE behind a public identity, and at least one UE is not in use, then the public identity is not considered busy since the operator cannot verify that the public identity is busy and, therefore, cannot interrupt the caller.

Information sufficient to bill the requesting party for the use of the BLV service must be collected, either by the operator services platform or by the PacketCable network, or both.

Busy Line Verification behaves as described in Telcordia LSSGR [GR 1176], but in the context of the PacketCable architecture. The feature implementation assumes an analog phone connected to a PacketCable-certified UE.

7.8.1.2 Feature Implementation

The operator services UE first places a test call to the target UE using basic call procedures as described in Section 7.1 to determine if the target UE is busy. If the target UE is busy, the operator services UE clears the call attempt using the basic call procedures as described in Section 7.1 and proceeds with the busy line verification procedure as described below.

After an initial test call has determined that the targeted line is busy, then the operator services position UE sends a SUBSCRIBE to the targeted public ID dialog event package with an Expires value of zero, which polls the target public ID's dialogs. The P-Asserted Identity in the SUBSCRIBE message MUST be the reserved operator ID, which is provisioned on both the operator position and on each subscriber UE via the BLV Operator Identity attribute as defined in Table 45.

If the Feature Availability Status attribute indicates the BLV feature is disabled, the target UE MUST respond to a SUBSCRIBE to the dialog event package from the operator BLV UE with a 403 Forbidden.

If the Feature Availability Status attribute indicates the BLV feature is enabled, the target UE MUST respond to a SUBSCRIBE to the dialog event package from the operator BLV UE with a 200 OK, followed by a NOTIFY message with a list of established dialogs at the UE. The UE SHOULD list the active dialog first.

Upon receiving the NOTIFY identifying the dialogs, the operator services UE MUST send an INVITE to the target UE with a JOIN header containing the first dialog ID in the NOTIFY. Upon receipt of the INVITE with a JOIN header identifying one of the active dialogs, the target UE MUST mix the RTP from the operator BLV UE and all existing dialogs after verifying that the P-Asserted ID in the INVITE is that of the operator. The mixed audio is sent to the operator UE. The audio is scrambled at the operator station prior to presentation to the operator.

If the operator determines that there is a call in progress, the operator can interrupt, per Section 7.8.2. If the requesting party declines the interrupt, or if the operator is incapable of interrupting the call, operator UE MUST send a BYE to the target public ID.

If the target UE is in a non-idle, non-talking state such as ringing or dialing, the target UE MUST reject the INVITE with JOIN header with 481 Call/Transaction Does Not Exist. When the operator services UE receives this 481 Call/Transaction Does Not Exist, it MUST play the reorder tone to the operator position.

If the target public identity no longer has an active dialog as indicated in the JOIN header within the INVITE, then the UE MUST respond to the INVITE with a 481 Call/Transaction Does Not Exist. If the target fails the JOIN request for any other reason, the UE SHOULD return a 488 Not Acceptable. If the JOIN request fails for any reason, the operator services UE MAY attempt an INVITE with a JOIN header identifying the next dialog ID on the list of dialog IDs in the original NOTIFY. If none of the JOIN attempts succeeds, the operator UE MUST play reorder tone to the operator.

If the NOTIFY did not indicate at least one active dialog, the operator services UE MUST send an INVITE without a JOIN header to the public identity of the target UE. If CFV is active at the public identity, the target UE MUST respond with a 486 Busy. If the public identity is idle without CFV active, the target UE MUST respond with 180 Ringing, and the operator UE MUST alert the operator with a ringback tone.

To minimize audio disruption in the call being verified, the UE SHOULD use the mixing services local to that UE rather than a network-based conference service. Wherever possible, the UE SHOULD avoid doing a reINVITE to other parties in the call to minimize audio disruption. A UE MAY reject an INVITE with JOIN header if it is already in a 3-way conference.

If the UE knows that it is in a TDD, FAX, or Modem call, the UE MUST answer the SDP with a=recvonly. The TDD, FAX, or Modem call between the users MUST NOT be disturbed because of BLV or Operator Interrupt.

An operator service uses a special trusted identity which is only allowed to be used for Busy Line Verification and Operator Interrupt. For security reasons, this identity MUST NOT be used by any UE other than the operator services UE. The UE MUST accept INVITEs with the JOIN header that uses this special identity unless specified otherwise in this section.

For a legacy based service, the operator service position attaches to the PacketCable network using a dedicated trunk circuit, called a No Test Trunk, which is attached to a Media Gateway. In this case, the MGC plays the role of the operator position UE as specified herein. See PacketCable [CMSS1.5] and [TGCP1.5] specifications for further information on how the MGC incorporates Busy Line Verification and Operator Interrupt related requirements.

If the target UE goes on-hook, the target UE MUST drop the BLV session by sending a BYE.

7.8.1.3 Feature Data

Table 45 summarizes the feature data defined to support the implementation of the Busy Line Verify feature.

Table 45 - Busy Line Verify Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Feature Availability Status	Boolean	Non-volatile	Per public user ID	XDS	XDS	UE	Mandatory
BLV Operator Identity (Note 1)	String	Non-volatile on XDS and HSS	network	XDS	XDS	UE, HSS, P-CSCSF, S-CSCF MGC, Operator UE	Mandatory
Note 1: It is recommended that this data element be stored in a secure manner on the customer UE (with limited access).							

7.8.1.4 Feature Interactions

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

Call Forwarding Variable

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

Do Not Disturb

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

7.8.1.5 Call Flows

The call flows are not normative, so if there is a discrepancy between the text requirements and these call flows, then the text requirements take precedence.

Figure 70 shows an example call flow for Busy Line Verification.

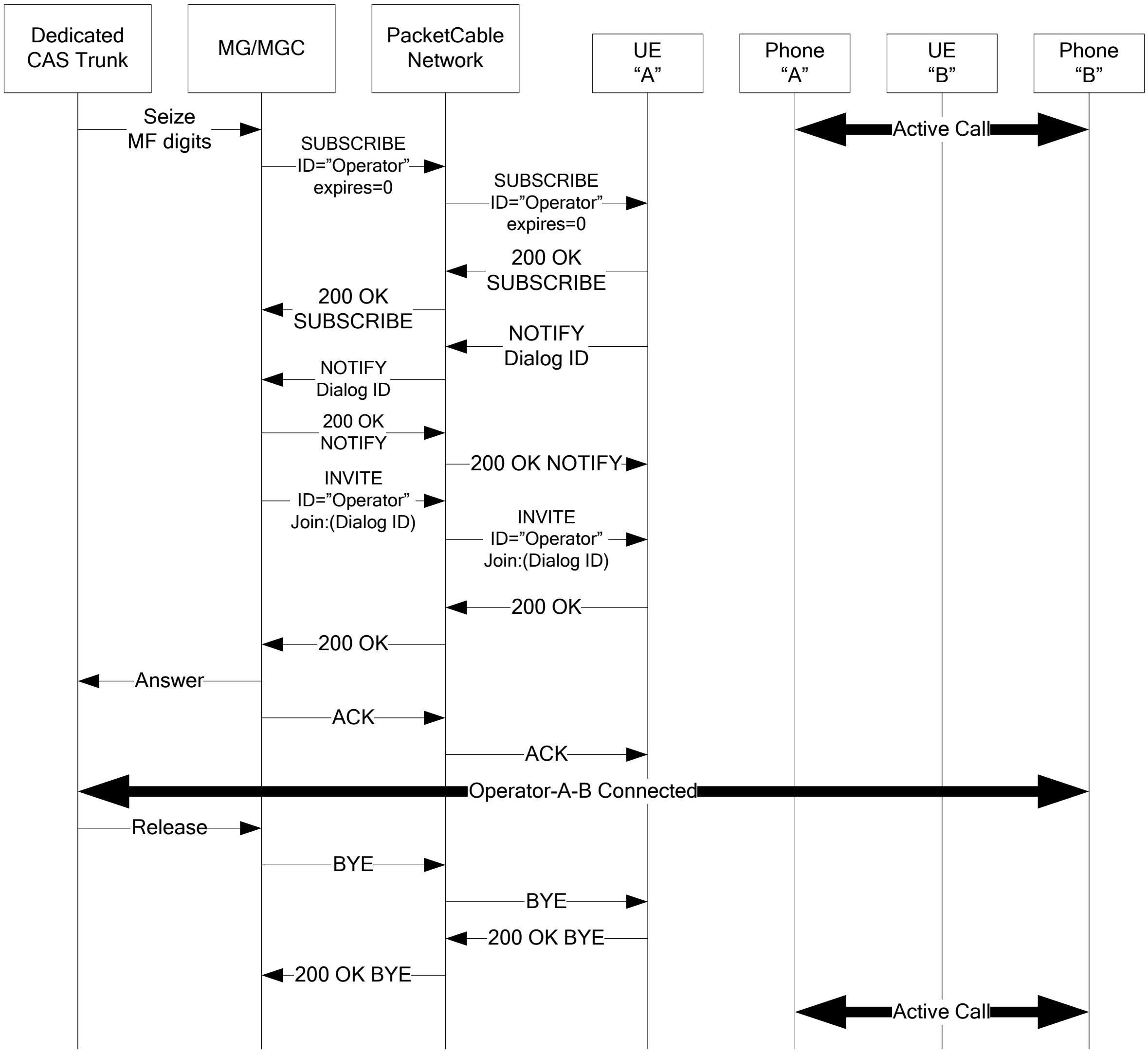

Figure 70 - Busy Line Verify (BLV)

7.8.2 Operator Interrupt

7.8.2.1 Feature Description

Operator Interrupt is performed as a follow-on step after the Busy Line Verification service. The requester asks the operator to interrupt the call. Operator Interrupt is allowable during 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. This provisioning is done through LIDB and the check of this provisioning is done by the operator services position. The UE doesn't need to know its provisioning status in this regard. If Operator Interrupt is provisioned as denied for the line being requested, the operator notifies the requester that Operator Interrupt is not possible. If the service 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 cut-through time for this intrusion is limited to 15 seconds. 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.

Information sufficient to bill the requesting party for the use of the Operator Interrupt and Call Completion services must be collected, either by the operator services platform or by the PacketCable network, or both.

7.8.2.2 Feature Implementation

Operator Interrupt implementation is identical to Busy Line Verification with the exception that the operator services position removes the scrambler from the receive path and takes the transmit path off of mute. From the point of view of PacketCable, the two features are identical.

7.8.2.3 Feature Interactions

Feature interactions for Operator Interrupt are identical to Busy Line Verification.

7.8.3 Call Completion

7.8.3.1 Feature Description

The Call Completion feature allows the operator services position to transfer a call from a requesting UE to a target UE. In the legacy circuit switched environment, the operator services position may keep the call on a local loop to monitor the call; or the operator services position may transfer the call using either Release Link Trunk methods for MF trunks or FACILITIES messages for Primary Rate ISDN or SS#7 signaling arrangements. In the case where the operator services position keeps the call on a local loop, the cross connect is made locally within the operator services position, and the PSTN sees two separate phone calls that terminate at the operator services position; one from the requesting UE to the operator and a second from the operator to the target UE.

7.8.3.2 Feature Implementation

The operator services UE MAY keep the call on a local loop connecting the requesting UE to the target UE using facilities local to the operator services UE. In this case, the operator services UE MUST use basic call procedures, as described in Section 7.1, to terminate the call from the requesting UE and initiate the second call to the target UE. The voice path between the two calls MUST be cross connected locally within the operator services UE.

The operator services UE MAY transfer the call to connect the requesting UE to the target UE. When transferring the call, the operator services UE MUST use the Call Transfer procedures described in 7.5.4.

The Operator Services position MAY be a legacy circuit switched service that attaches to the PacketCable network using a dedicated trunk circuit attached to a Media Gateway. In this case, the Media Gateway Controller MUST play the role of the operator position UE as specified herein.

8 EMERGENCY SERVICES

This section provides a technical overview and specifies requirements for the support of emergency services in PacketCable. The requirements are based on the referenced IETF Internet-Drafts and emerging documents from NENA.

Support for emergency services in a PacketCable network involves multiple network elements, specifically the P-CSCF, E-CSCF, MGC, SG, and the MG.

8.1 Overview of NENA E9-1-1 VoIP Migration Phases

NENA is an organization which has a mission of fostering the technological advancement, availability, and implementation of a universal emergency telephone number system (9-1-1) in North America. NENA has created a VoIP/Packet Technical Committee to define how 9-1-1 calls that originate on a VoIP service provider network should be handled. This committee has specified an implementation plan in three phases based on interim architectures:

- **i1:** NENA i1 documents mechanisms in use today to support 9-1-1 services by some VoIP service providers. [NENA i1] allows for VoIP 9-1-1 calls to be delivered to the correct PSAP via a PSAP-designated 10-digit number.
- **i2:** NENA i2 defines a transition mechanism that is roughly based on the cellular 9-1-1 process to allow a VoIP Service Provider to terminate VoIP calls to a 9-1-1 operator with an automatic location and without requiring "Public Safety Answering Points" (PSAPs) to upgrade their current systems [NENA i2].

  An intermediary phase, termed "Pre i2" has also been introduced. NENA pre-i2 allows that a VoIP 9-1-1 call be delivered to the correct PSAP via dedicated access to the Selective Router, which then routes the call to the appropriate PSAP.
- **i3:** NENA i3 defines a new, all IP-based architecture for PSAPs that allows VoIP emergency calls to be terminated as VoIP directly at the PSAP (requiring network upgrades for PSAPs) [NENA i3].

This section specifies the technical requirements for the support of NENA's Interim VoIP architectures for Enhanced 9-1-1- Services, i1 [NENA i1], Pre i2, i2 [NENA i2], and i3 [NENA i3].

8.2 Scope

The following requirements are considered to be in scope for PacketCable Emergency Services Support:

- Identifying the location of the UE and storage of location information by the UE.
- Identifying emergency calls at the UE and/or CSCF servers.
- Conveying location information in the SIP signaling for emergency calls, following the IETF protocol mechanisms.
- Special handling of emergency calls, including establishing QoS and priority treatment for emergency call media and signaling, and special SIP signaling prioritization processing or marking. QoS and priority treatment are not addressed in this version of the specification.
- Handling of emergency calls at SIP Proxies and other PacketCable routing server functions.

8.3 Assumptions

The following are the assumptions made in support of PacketCable Emergency Services.

- Given the scope of PacketCable, the on-going work in 3GPP IMS on emergency services, and the importance of E-911 support in VoIP networks, the applicability of the [NENA i1], pre-i2, i2, and i3 interim phases for North America is assumed.
- The use of IETF protocols and methodologies for location determination and conveyance is assumed; and at the time of this writing, the approach is in line with the work-in-progress in 3GPP.
- On the network side, this specification assumes that location determination on access networks where UEs may roam is available, including location determination on DOCSIS cable access networks.
- This specification assumes that location information is provided to the UE via DHCP. The mechanisms for providing location information on DOCSIS access networks may vary and the detection of a location change on cable networks are out of scope of this specification.

  For UEs operating behind local DHCP servers (for example, home routers with NAT capabilities), the location information DHCP option should be relayed. In the absence of a DHCP relay for the location information, a dynamic location may not be reliably provided by the UE, and the E-CSCF handles the 911 call without location and attempts to do a static mapping.
- The UE supports the DHCP protocol and the associated DHCP options for geographical location [RFC 3825] and civic location [ID DHCP-CIVIC]. The UE also supports SIP multipart MIME as specified in [RFC 3261] to convey location information in SIP message bodies as defined below.

8.4 Emergency Call Handling

This informative section describes the basic steps in emergency call handling in [NENA i1], [NENA i2], and [NENA i3] environments. They include:

1. At boot time, the UE requests its location from the access network using DHCP. The UE must request the DHCP Option for Civic Addresses Configuration Information [ID DHCP-CIVIC] by including the option 99 in the DHCP 'parameter request list' option value (DHCP option 55) of the broadcast DHCPv4 DISCOVER message in IPv4.

   In IPv6, the UE must use DHCPv6 [RFC 3315] to request its location from the access network. The UE must include an Option Request option (OPTION_ORO) with a requested-option-code value of 99 (GEOCONF_CIVIC) in a Solicit, Request, Renew, Rebind, Confirm or Information-request message in order to inform the DHCPv6 server about DHCP civic option the client wants the server to send to the client.

   Other configuration parameters related to emergency services support may also be provided to the UE by the provisioning systems such as the local emergency dialing strings (for example, 9-1-1).
2. The user indicates the local dialing string for emergency calling (for this example, it is assumed 9-1-1 is indicated or dialed on the phone).

   PacketCable uses "the user indicates" as opposed to the user dials the local dialing string to be more generic. It is defined to follow whatever the user picks as the specific dial string to be used for emergencies, whether the dialing string is selected by pressing numeric keypad buttons to push in a certain sequence, by picking the 911 sequence from a menu on a display of the phone, or by merely selecting an "emergency" button — which could be hardware or software-based — as the number selected for this call. The UE performs digit analysis and determines that this is an emergency call based on configuration data.
3. The UE initiates a SIP INVITE transaction with the universal emergency service URN "URN:service:sos" defined in [ID SIP-URN] as the Request-URI. The To: header value of the SIP INVITE request is also the "URN:service:sos."

In some countries outside North America, the UE may be configured to support multiple local emergency dial strings. The dial plan configuration mechanism defined in PacketCable provides the means to translate specific dial strings into specific URIs or service URN (for example, the 112 dial string may translate into URN:service:sos and 18 may translate into URN:service:sos.fire).

4. The INVITE request also includes the location obtained from DHCP in its message body in the form of a Presence Information Data Format Location Object (PIDF-LO) [RFC 4119] as defined in the SIP Location Conveyance specification [ID SIP-CONVEY].

   The UE does not include the route received in the route header received at registration. However, the UE may include the universal emergency service URN "URN:service:sos" in the Route header and use a previously received PSAP URI in the Request-URI.

5. The P-CSCF detects the emergency call based on the presence of the "URN:service:sos" URN and forwards the call to an Emergency-CSCF (E-CSCF).

   The E-CSCF and P-CSCF are described as logical functions; the E-CSCF functions such as those described in steps 6-9 may be implemented in the same physical network element as a P-CSCF, a S-CSCF, or in a separate device.

6. The E-CSCF then follows the procedures outlined below, depending on the supported interim NENA architecture, which is [NENA i1], pre-i2, i2 [NENA i2], or i3 [NENA i3] compliant.

NENA i1 Architecture

The E-CSCF must convert the SIP Request URI into a 10-digit emergency services number. How this conversion is done is beyond the scope of this specification.

The call is routed to the PSTN via the MGC. In this scenario, the call does not have location provided to the PSAP through mechanisms defined in this specification.

NENA pre-i2 Architecture

The call is routed to the PSTN via an MGC connected to the E911 selective router. In this scenario, the caller's telephone number is used to look up the UE location in the selective router for routing to the proper PSAP.

NENA i2 Architecture

The E-CSCF may act as an emergency services routing proxy, as defined in [NENA i2]; in this case it relies on PSTN interfaces for terminating the call to the PSAP using the PSTN. When the E-CSCF acts as a routing proxy following the [NENA i2] requirements, the following applies:

- The E-CSCF extracts the location of the caller from the INVITE message body.
- The E-CSCF queries a VoIP Positioning Center (VPC) providing the location of the caller.
- The VPC returns a routing code or Emergency Service Routing Number (ESRN) and an Emergency Services Querying Key (ESQK) that can be used for routing the call.
- The E-CSCF forwards the call by initiating an INVITE transaction with the ESQK in the appropriate header (either the P-Asserted-Identity [RFC 3325], the P-Preferred-Identity, or the From) and the ESRN in the Request-URI as defined by NENA.
- The call is routed to an Emergency Services Gateway (ESGW). The ESGW is the NENA functional element that maps the given ESQK and ESRN to the PacketCable Signaling Gateway (SG), Media Gateway Controller (MGC), and Media Gateway (see Section 8.8.2)
- The ESGW (as implemented by the SG, MGC, and MG) places the call on the appropriate Selective Router and provides the ESQK as the calling party's telephone number.

- The Selective Router routes the call to the correct PSAP based on the calling party's telephone number (ESQK).
- The PSAP answers the call. It queries the Automatic Location Indication (ALI) database with the ESQK it receives as the calling party's telephone number.
- The ALI database steers the query to the VPC that provided the ESQK.
- The VPC returns the location of the caller and its callback number.
- The data is returned from the ALI database and is presented to the PSAP.

NENA i3 Architecture

- The E-CSCF extracts the location of the caller from the INVITE message body.
- The E-CSCF uses a static mapping or a mapping protocol to obtain the URI of the PSAP servicing the location of the caller.
- If a valid PSAP URI is obtained from the mapping query, the call is forwarded by the E-CSCF to the PSAP SIP server. The IP-enabled PSAP (NG9-1-1 PSAP) accepts the incoming call directly as a SIP session.
- If a PSAP URI is not returned from the mapping protocol because the PSAP does not accept SIP calls, or if the mapping fails, the E-CSCF may proceed with [NENA i1], pre-i2, or i2 [NENA i2] as indicated in step 6.

8.5 UE Requirements for Emergency Services Support

This section defines the minimum set of requirements for a UE to be capable of supporting emergency services in PacketCable networks.

8.5.1 Minimum UE State to Place Emergency Calls

A UE MUST, at minimum, meet all the following conditions in order to be able to successfully make an emergency call:

- The UE is capable of attempting an emergency call.
- The UE has obtained the domain name of a PacketCable service provider.
- The UE can establish connectivity with the SIP network.
- The UE is participating in a keep-alive message exchange with the PacketCable network after obtaining the emergency call configuration.
- The UE is registered and authenticated before an emergency call is placed.

A UE in the minimum state defined above should be returning a dial tone for a telephone to which it is connected. If the UE is presenting the dial tone, 9-1-1 calls can be placed.

8.5.2 Emergency Calling Configuration

The following configuration parameters MUST be provided to a UE in support of emergency calling:

- A digit map identifying the sub-set of dialed numbers that must be recognized as emergency numbers.
- The UE's location information: a set of location parameters obtained via DHCP.
- The Request-URI to be used for emergency services, the default value being "urn:service:sos."

- The preference for the type of location format (Presence Information Data Format - Location Object, or PIDF-LO) the UE should include if it gets both civic and geographic location information via DHCP.
- Other configuration parameters for special prioritization of media and signaling traffic of emergency calls as defined in 8.5.9 and 8.5.10.

The UE MUST be capable of handling 911 calls if dial-tone is provided, even when one or more of the above configuration parameters are not provided. For example, the UE may handle the call without prioritization or media QoS if the associated configuration parameters are not provided, but it MUST attempt to place the 9-1-1 call.

8.5.3 Obtain Location

The UE is responsible for obtaining its location. A UE MUST be capable of receiving its location via DHCP by processing the DHCP options that provide the civic address or coordinate-based (geographic) location configuration information.

A UE MUST support the GEOCONF CIVIC DHCP option defined in the [ID DHCP-CIVIC], and the UE SHOULD also support GeoConf DHCP option 123 defined in [RFC 3825]. A UE MUST request the GEOCONF CIVIC DHCP option in DHCP option 55 in IPv4, and in the Option Request Option in DHCPv6. If both the DHCP GeoConf and GEOCONF CIVIC options are supported, the UE MUST indicate or request both options in DHCP.

The resulting location information is then used by the UE to create a PIDF-LO using the mechanism described in [RFC 4119] and illustrated in [RFC 4119]. The UE MUST use the value of the GEOCONF CIVIC DHCP option to generate a PIDF-LO if both DHCP options 123 and 99 are returned by the DHCP server.

The UE MUST store the location information, along with the timestamp of when it receives its location information, via DHCP. The PIDF-LO generated by the UE from the DHCP location information object MUST contain a timestamp element. The value of the timestamp MUST be the time when the UE receives its location information.

When the UE registers, it SHOULD indicate support for the location, as defined in the SIP Location Conveyance specification [ID SIP-CONVEY]. In particular, the UE MUST support SIP multipart MIME, as specified in [RFC 3261], to convey location information in SIP message bodies.

8.5.4 Recognition of an UE-Originated Emergency Call

The recognition of an emergency call is a critical step in assuring its proper handling and call routing. Technical requirements are, therefore, placed on both the UE and the P-CSCF to perform such recognition.

The UE-based digit map action MUST be provisioned with an indication to invoke the EMERGENCY-CALL digit map action upon successful entry of a dial string configured as an emergency number as defined in Section 7.1. Upon invoking EMERGENCY-CALL digit map action, the UE MUST proceed with the UE requirements identified in the remainder of this Section 8.5.4, and continue with the other UE requirement in Sections 8.5.5 through 8.5.11.

The UE MUST recognize the local dialing string for emergency calling, through analysis of the digit map, that the user has dialed (for example, 9-1-1 in the United States) based on the digit map configuration. Special processing of the call should immediately take place.

In addition, the UE MUST recognize standard North American forms of prefix dialing in front of the emergency calling digits based on digit map configuration, including digit maps such as 0+911, 1+911, and 01xxx+911. The UE must be able to map several dialed strings or digit maps to the emergency service feature and its associated service URN.

8.5.5 Basic Call Modifications while on an Emergency Call

Upon determining that an emergency call has been initiated (see Section 8.5.4), the UE and the associated network elements involved in the emergency call MUST observe a number of specific call interaction requirements, as defined in Section 8.5.5.

The following section illustrates a scenario in which a UE is controlling an analog phone. A sample of an informative basic call is shown in Figure 71:

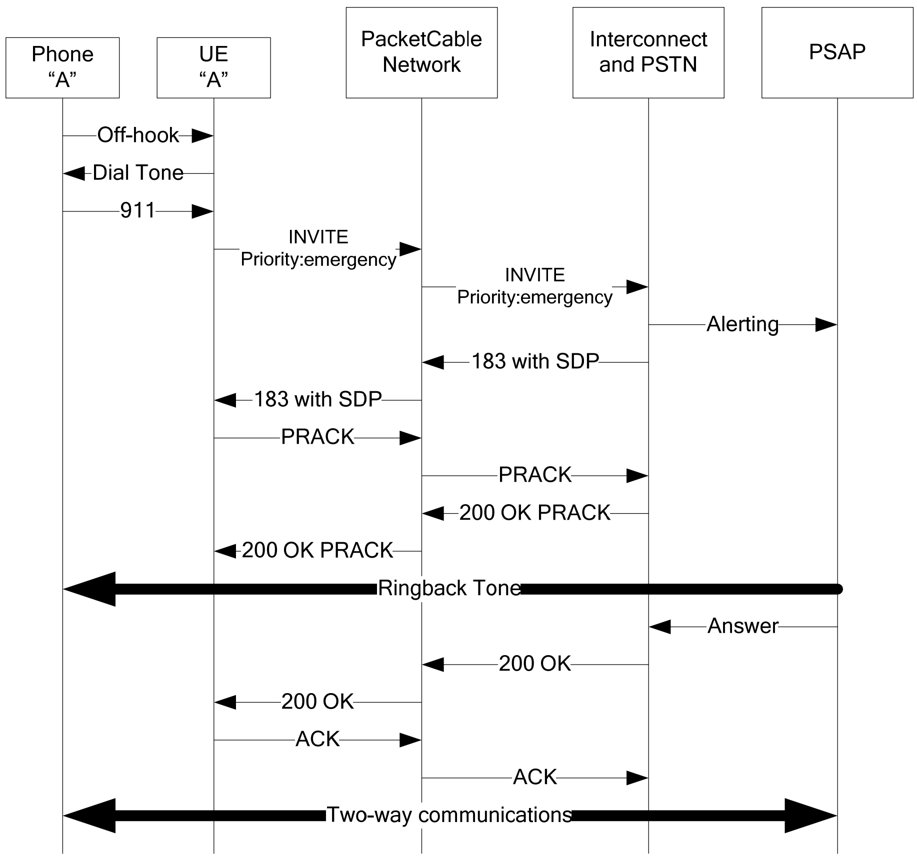

Figure 71 - Emergency Call

8.5.5.1 Basic Call – SIP INVITE to initiate an Emergency Call

The UE generates a SIP request to initiate a call and sends it to the P-CSCF. The SIP INVITE request initiating an emergency call MUST have:

- The Request-URI set to the universal emergency service URN "URN:service:sos" as defined in [ID SIP-URN].
- The From: set to the user identity of the caller (as normal in PacketCable).
- The To: set to emergency service URN "URN:service:sos."
- A multi-part MIME message body including:
  - The location as a PIDF-LO location object if the location is provided to the UE. If no current location is available, the Location header MUST include the option tag value of 'Unknown-Location' as described in the SIP Location Conveyance specification [ID SIP-CONVEY].

    - The SDP message body part. The ITU-T G.711 codec MUST be offered in the media line attribute. If a wideband codec is supported by the UE, it MAY be offered as an additional choice. The UE MUST NOT offer any other codecs. Voice Activity Detection SHOULD be disabled.

- Any other mandatory SIP headers as defined in [PKT 24.229].
- SIP signaling identification of an emergency call as described in Section 8.5.9.

The UE SHOULD include a language preference whenever possible, according to the procedures in Section 7.1.6.8.

Other optional SIP headers MAY be provided, including Route headers with pre-calculated mapping of the location to the PSAP URI as a loose route.

If a call is returned to the UE with a 424 (Bad Location Information) response code (see [ID SIP-CONVEY]), the call should be reattempted as an emergency call per Section 8.5.1. Upon receipt of a 424 (Bad Location Information) response code, the UE SHOULD retry to determine its location using the method previously used (DHCP, for example) and it SHOULD make a new attempt to initiation the emergency call. After the third unsuccessful attempt (with the same location information in the SIP message body), the UE MUST stop retrying.

8.5.5.2 Basic Call – Early Media

No special interactions are noted between early media and emergency calling. The UE does not behave any differently for early media (see Section 7.1.9) if the call is an emergency call.

8.5.5.3 Basic Call – Terminating Call

When the UE is in an emergency call, the UE MUST reject all incoming calls that are not emergency calls with a 486 – Busy here response.

The UE MUST accept a ringback request from the PSAP, which is signaled using a RE-INVITE with a special ringback Alert-Info header. When the UE receives the RE-INVITE with the special ringback Alert-Info header, it MUST generate ringing (if the UE is on-hook) or a configured duration howler tone (if the UE is off-hook).

Note: If the UE hangs up while receiving a howler tone due to ringback, the howler tone stops, the operator position is notified through a reinvite with hold SDP, and the operator is notified that the phone is on-hook by receiving low tone at his or her position, and may then choose to cause ringback again.

The UE MUST accept a callback request from the PSAP.

8.5.5.4 Basic Call - Hold

The UE MUST NOT initiate a HOLD (see Section 7.1.5) while active on an emergency call.

While active on an emergency call, the UE MUST be capable of receiving a hold request (see Section 7.1.5).

8.5.5.5 Basic Call – No Answer Timeout

No special interactions are noted here.

8.5.5.6 Basic Call – Response Codes and RE-INVITE

The UE MUST NOT initiate a Redirect (see Section 7.1.11) while active on an emergency call. Additionally, the UE SHOULD NOT send a 300 Multiple Choices, 301 Moved Permanently, or 302 Moved Temporarily response to an emergency call ringback or callback.

While active on an emergency call, the UE MUST be capable of receiving a redirect request (see Section 7.1.11), since this is often done to transfer the emergency call to the proper agency to provide assistance.

8.5.5.7 Basic Call – Forced Disconnect

The UE MUST honor a BYE request from PSAP on an emergency call (PSAP forced disconnect); this ensures that the resources used for the emergency call are released for the next emergency call. Although this is normal UE behavior, no feature or service should allow this behavior to be modified while on an emergency call.

8.5.5.8 Basic Call – Network Hold Support

Network Hold is an optional feature that provides the ability to hold the network connections for the 911 call in the event that the caller tries to release the session prior to disconnecting from the PSAP operator. If Network Hold is supported in the network in the PSTN, the PSAP operator hears a tone to indicate that a caller has gone on-hook and the hold has been activated.

The session and the call resources are held in the network until either:

- The PSAP operator releases the call (forced disconnect).
- The UE reconnects to the original emergency call session.
- The UE network hold timer expires.

Because only the PSAP knows if the network hold feature is in effect, the UE must assume that network hold could be applied. This means that the UE processing of a disconnect request from the user differs from standard basic call processing of a user disconnect after the PSAP answers.

If the UE disconnects before receiving a 200 OK (answer) on the emergency call, the UE MUST process that disconnect as if this is not an emergency call, and send according to [RFC 3261].

Upon receiving a disconnect from the user while on an emergency call, and after receiving the first 200 OK from the PSAP, the UE MUST send a hold event to the network (see Section 7.1.5) and start a Network Hold timer. The network hold event sent to the network MUST contain the emergency call indicator headers so that the network can differentiate that hold from a user initiated hold, per Section 8.5.5.4, which should be denied. This is shown in Figure 72.

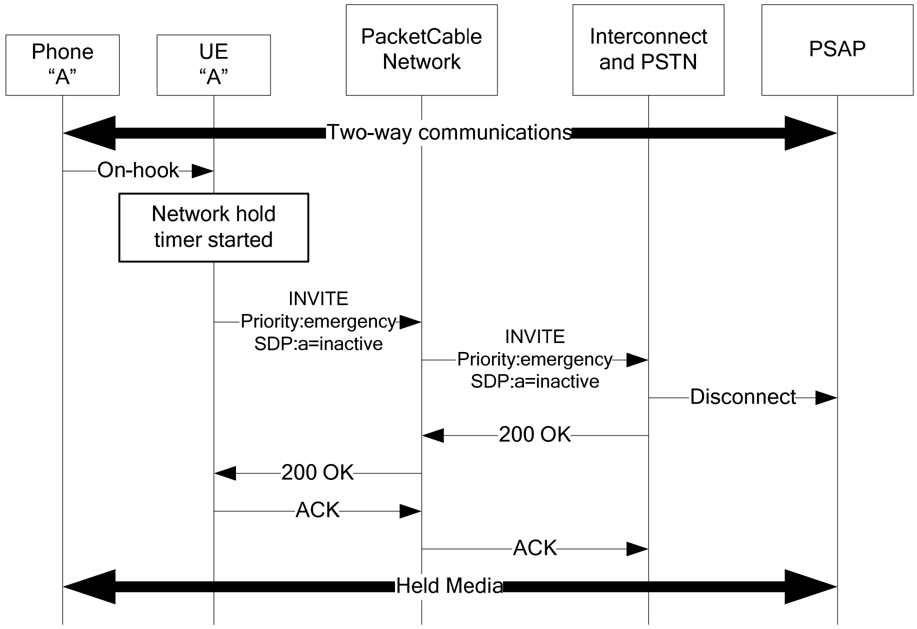

Figure 72 - Emergency Call; User disconnect handling

If the user attempts to reconnect to the call, then the UE MUST return from hold (see Section 7.1.5), reconnect with the PSAP, and cancel the network hold timer at that time. This process is shown in Figure 73.

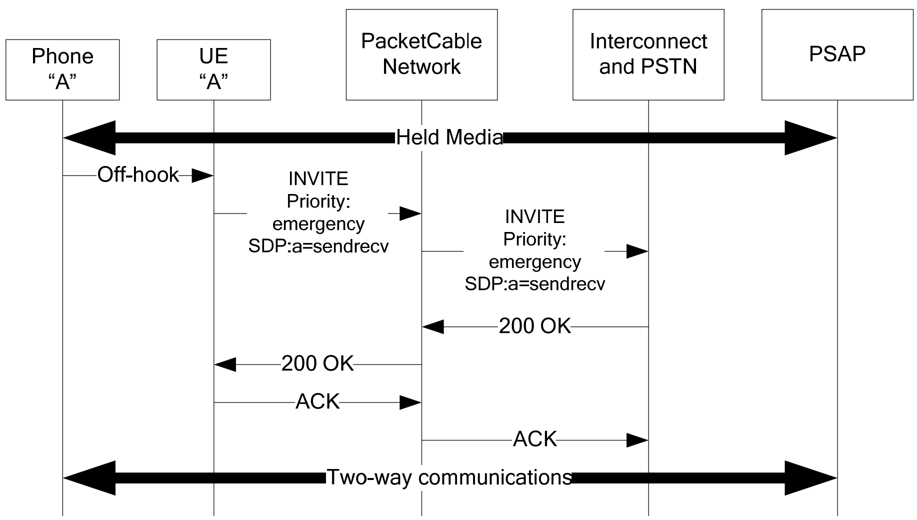

Figure 73 - Emergency Call; Reconnecting after hold

When the UE network hold timer expires, the UE MUST send a BYE towards the network to release the network resources.

The value of the network hold timer should be provisioned on the UE (see Table 46). A value of '0' means that the network does not support network hold, although this is not recommended unless it is known that all PSAPs reachable from this UE do not support network hold.

8.5.6 PSAP Operator RingBack

Operator Ringback is a feature that allows the PSAP to alert the caller. The type of alert depends on whether there is an established bearer path to the caller or not. If there is an established bearer path (the caller is offhook), the caller can be alerted with a special in-band audible alerting (usually a ROH tone). If the caller is on Network Hold (onhook), then the caller is alerted with standard power ringing alerting. While the caller is receiving the ringback, the PSAP should be receiving audible alerting.

8.5.6.1 PSAP Ringback with an established bearer path (Offhook Caller)

When the PSAP indicates that the caller should be alerted and the caller has an established bearer path, the network sends the caller a special in-band audible alerting in the standard bearer path. This behavior is not specific to the fact that this is an emergency call. No unique behavior is required by the UE for emergency calls.

The network sends ROH through a REINVITE of the bearer path to a server that can provide the special in-band alerting. The UE MUST accept REINVITEs per Section 8.5.5.6. The length of tone is decided by the network and the network provides the audible ringing towards the PSAP.

Alternatively, the UE MAY receive a request to play a tone (per [RFC 2833], usually ROH). In this circumstance, the UE MUST play back that tone for the length of time specified in the request. UEs must support reception of the ROH tone, per [RFC 2833]. The network is responsible for providing audible ringing towards the PSAP.

If the UE receives an onhook indication while playing ringback, the UE MUST behave as if ROH is not playing and send a hold event according to Section 8.5.5.8.

8.5.6.2 PSAP Operator Ringback while on Network Hold (Onhook Caller)

When the PSAP indicates that the caller should be alerted while the caller is on network hold, the UE should continue to run the Network Hold timer and provide the physical ringing indicated. This is shown in Figure 74.

To accomplish the ringing, the network sends a REINVITE with an Alert-Info header that indicates the type of physical alerting that should be provided. The UE MUST provide the physical alerting specified. If the UE does not support the alert pattern specified in the Alert-Info header, or if it cannot retrieve the alert pattern, then the UE MUST substitute standard ringing. The UE processes this invite according to basic call procedures, including sending of 180 ringing and responding as appropriate for a standard terminating call. Additionally, the UE MUST set the emergency call indicator headers.

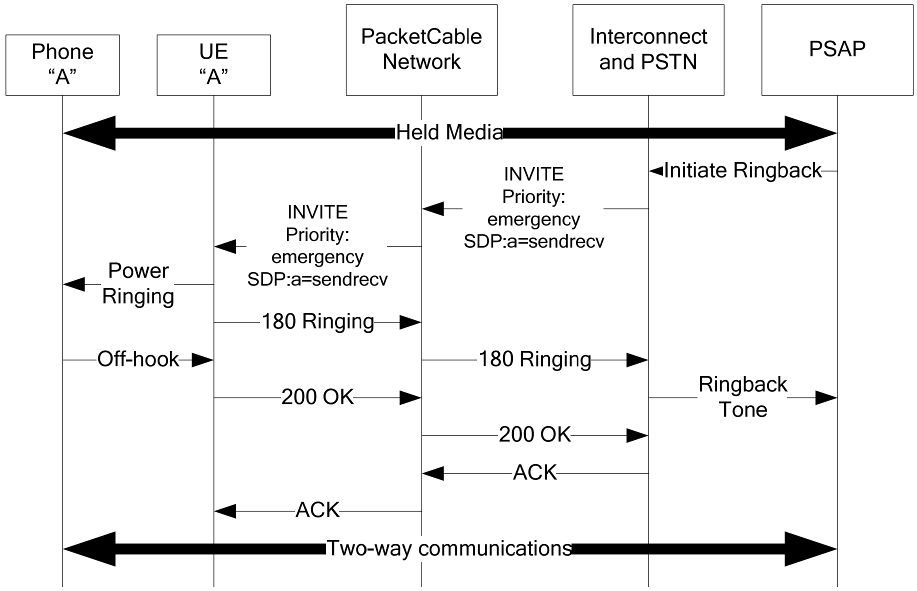

Figure 74 - Emergency Call; PSAP operator ringback while on hold

The network provides audible ringing back to the PSAP until it receives a 200 OK from the UE. If the UE receives a BYE from the PSAP during the alerting, it SHOULD send a CANCEL or BYE to the network as appropriate (PSAP forced disconnect).

If the Network Hold timer expires during the Operator Ringback procedure, the UE MUST send a CANCEL or BYE towards the network as appropriate.

If the caller answers, the UE MUST send a 200 OK according to basic call procedures and cancel the Network Hold timer.

8.5.7 PSAP CallBack (PSAP Originated Emergency Call)

After an emergency call is ended (due to a caller-initiated disconnect before 200 OK or a PSAP forced disconnect), the PSAP operator may still wish to re-establish the connection with the caller. In this case, the PSAP initiates a new call towards the UE, which can be identified as an operator-initiated emergency callback by the UE, based on the presence of the well-known network asserted identity that matches the sub-service 'sos' as defined in [ID SIP-URN].

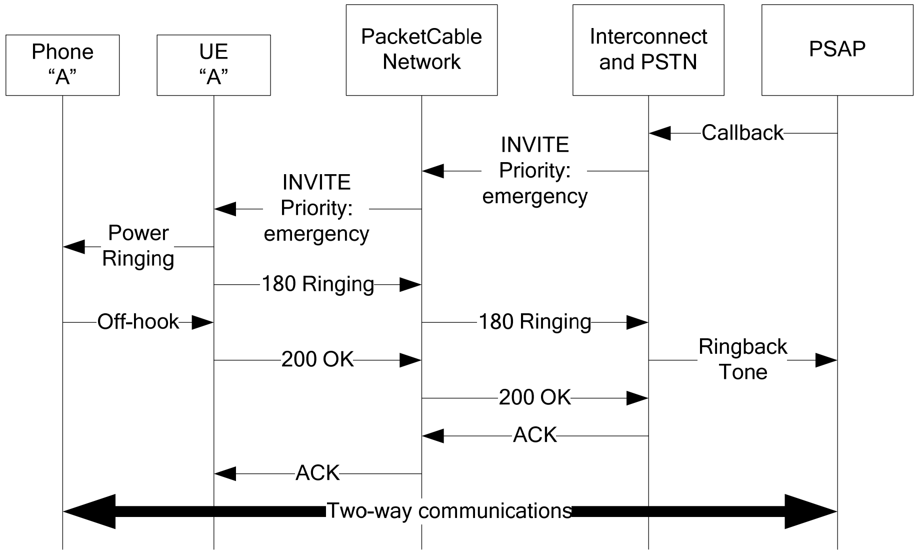

Figure 75 - Emergency Call; PSAP operator callback

In this case, the UE MUST consider the incoming request as an emergency call initiation and behave as if the UE initiated the emergency call. That is, the basic call procedures in 8.5.5, the Network Hold Procedures in Section 8.5.5.8, the PSAP Ringback procedures defined in Section 8.5.6, and the feature interaction procedures defined in Section 8.5.8 MUST all still apply. This is shown in Figure 75.

8.5.8 Feature Interactions

Upon determining that an emergency call has been placed (see Section 8.5.4), the UE involved in the emergency call conforms to a number of specific feature interaction requirements, as defined in this section.

Caller ID Delivery

No interactions. Feature should work as specified.

Caller ID Per-Line Blocking

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

Caller ID Per-Call Blocking

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

Caller ID Per-Call Delivery

No interactions. Feature should work as specified.

Anonymous Call Rejection

No interactions. Feature should work as specified. PSAP ringbacks and callbacks are not anonymous.

Call Forwarding Variable

The UE that provides an analog telephone interface MUST implement the feature interactions between Call Forwarding Variable and emergency services as specified in Telcordia [GR 529] and [GR 580].

The UE MUST NOT forward an emergency ringback and emergency callback call. The CFV AS MUST NOT allow programming of a redirection service to terminate on the emergency services TEL or SIP URI.

Call Forwarding Do Not Answer

The UE that provides an analog telephone interface MUST implement the feature interactions between Call Forwarding Do Not Answer and emergency services as specified in Telcordia [GR 529] and [GR 586].

The UE MUST NOT forward an emergency ringback (RE-INVITE) and emergency callback call. If, while the UE is in an emergency session, a UE receives an INVITE that does not contain the emergency call indications, then the UE MUST reject the call with 486 – Busy here. The UE MUST NOT allow programming of a redirection service to terminate on the emergency services TEL or SIP URI.

Call Forwarding on Busy

The UE that provides an analog telephone interface MUST implement the feature interactions between Call Forwarding on Busy and emergency services as specified in [GR 529] and [GR 586].

The UE MUST NOT forward an emergency ringback (RE-INVITE) and emergency callback call. The UE MUST NOT allow programming of a redirection service to terminate on the emergency services TEL or SIP URI.

Selective Call Forwarding

The UE that provides an analog telephone interface MUST implement the feature interactions between Selective Call Forwarding and emergency services, as specified in [GR 529] and [GR 217].

The UE MUST NOT forward an emergency ringback (RE-INVITE) and emergency callback call. The UE MUST NOT allow programming of a redirection service to terminate on the emergency services TEL or SIP URI.

Remote Activation of Call Forwarding

The UE MUST NOT allow programming of a redirection service to terminate on the emergency services TEL or SIP URI.

Outbound Call Blocking

Call blocking service to emergency services should not be allowed when the service allows the dial-tone to be provided.

Collect Call Blocking

Collect calls to emergency services should not be blocked.

900/976 Call Blocking

No interactions. Feature should work as specified.

Call Waiting

The UE that provides an analog telephone interface MUST implement the emergency call interactions according to [GR 529] and [GR 571].

Additionally, all incoming calls should be blocked unless they include the emergency call indicators. When a UE receives a call without the emergency services indicator, it MUST respond with a 486 – Busy Here.

Hold

The UE that provides an analog telephone interface MUST implement the emergency call interactions according to [GR 529], [GR 577] and [GR 579].

The UE MUST NOT put an emergency call on hold.

Call Transfer

The UE that provides an analog telephone interface MUST implement the emergency call interactions according to Telcordia [GR 529] and [GR 577].

The UE MUST NOT transfer an emergency call.

Three Way Calling

The UE that provides an analog telephone interface MUST implement the emergency call interactions according to Telcordia [GR 529] and [GR 577].

The UE MUST NOT invoke a three-way calling feature in an emergency call.

Do Not Disturb

The UE that provides an analog telephone interface MUST implement the emergency call interactions according to Telcordia [GR 529].

The UE SHOULD NOT reject an emergency ringback (RE-INVITE) and an emergency callback call.

Subscriber Programmable PIN

No interactions. Feature should work as specified.

Distinctive Alerting

No interactions. Feature should work as specified.

Message Waiting Indicators

The UE that provides an analog telephone interface interrupts an MWI transmission for an incoming call. Therefore, the PSAP ringback and callback take precedence over MWI. The MWI is applied after the current call is completed per Section 7.6.4

Speed Dialing

The Speed Dial AS MUST implement the emergency call interactions according to Telcordia [GR 570].

Customer-Originated Call Trace

The UE that provides an analog telephone interface MUST implement the emergency call interactions according to Telcordia [GR 570] and [GR 216].

Additionally, the UE and the COT AS MUST NOT allow an emergency ringback or callback to be traced.

Screen List Editing (SLE)

The UE MUST NOT allow programming of a SLE service with the emergency services TEL or SIP URI.

Auto Recall

The UE that provides an analog telephone interface MUST implement the emergency call interactions according to Telcordia [GR 227].

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

Auto Callback

The UE that provides an analog telephone interface MUST implement the emergency call interactions according to Telcordia [GR 2948] and [GR 215].

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

Busy Line Verification

No interactions. Feature should work as specified. BLV of an emergency services call is allowed.

Emergency Interrupt

No interactions. Feature should work as specified. Emergency Interrupt of an emergency services call is allowed.

Call Completion

No interactions. Feature should work as specified. Call Completion of an emergency services call is allowed.

8.5.9 Signaling Identification of an Emergency Call

Once a call has been determined to be an emergency call (see Sections 8.5.4 and 8.6.1), additional requirements are applied to the signaling to ensure proper priority of the call and the signaling through the use of additional SIP headers [RFC 4412].

8.5.9.1 SIP Priority Header Marking of an Emergency Call

The SIP Priority header is defined in [RFC 3261] and provides a user indication of the importance that the SIP request should have for the receiving human or user agent. The 'Priority' header field does not affect the usage of PSTN gateway or proxy resources. The UE MUST set the SIP Priority header for emergency session establishment.

Example:

```
Priority: emergency
```

Note: The SIP Resource-Priority Header is defined in [RFC 4412]. Its use for emergency calls is not specified and is left for further study. Network elements SHOULD NOT require support for the SIP Resource-Priority header for emergency calls: it is NOT RECOMMENDED to use the SIP Resource-Priority option tag in a SIP Require header for this purpose of emergency calls.

8.5.10 Media QoS for an Emergency Call

The UE MUST mark the media packets of emergency calls with special DSCP values based on the UE configuration parameters [RST PACM] and name those parameters).

8.5.11 Other UE Requirements

In the INVITE sent by the UE to the P-CSCF for emergency call, the UE MUST NOT include the route received in the route header that was received in the registration response.

8.5.12 Emergency Services Feature Data

Table 46 summarizes the feature data elements defined to support the implementation of the Emergency Services feature.

Table 46 - Emergency Services Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	PACM Requirement
Network Hold Timer	Integer (minutes) (default: 45; see Note 1)	Non-volatile	Per network	XDS	XDS	UE	Mandatory
Emergency Howler Duration	Integer (seconds) (default: 3)	Non-volatile on XDS, volatile on UE	Per Network	XDS	XDS	UE	Mandatory
Emergency Media DSCP Value	Integer (range of 0..63)	Volatile in UE, non-volatile in XDS	Per Network	XDS	XDS	UE	Mandatory
Note 1: Default is set according to [GR 529].							

8.6 P-CSCF Requirements

8.6.1 Recognition of UE-Originated Emergency Call

The P-CSCF MUST recognize a SIP request with a Request-URI of "URN:service:sos" as a session initiation for an emergency call.

In the event that, for some reason, the UE does not recognize that the local dialing string for emergency calling has been dialed (for example, 9-1-1), the P-CSCF MUST recognize that the emergency dialed number and the P-CSCF SHOULD immediately reject the call with a 424 (Bad Location Information) response code as defined in the SIP Location Conveyance specification, [ID SIP-CONVEY]. In order to support non-UEs, upon receipt of a second SIP request for an emergency dialed number from the same UE, the P-CSCF may accept the call and process the request accordingly.

Testing of the SOS URN SHOULD be accomplished immediately upon receipt of the message and before other non-emergency processing so as to maximize the possibility that emergency calls go through.

The P-CSCF MUST also perform a special check for dialog initiating requests containing a Request-URI with a TEL or SIP URI formed with a user part containing a local emergency dial string (for example, tel:911 or sip:911@msodomain.net;user=phone). This case occurs when the UE does not recognize the call as an emergency call. In this case, the P-CSCF MUST change the Request-URI to the emergency services URN "URN:service:sos" and proceed as described in Section 8.6.2; the location may not be provided by the UE.

8.6.2 Forwarding the call to an E-CSCF

Upon determination that the call attempt is an emergency call, the P-CSCF MUST forward the call to an E-CSCF on the same network as the P-CSCF.

The P-CSCF SHOULD check for the presence of the SIP Priority header according to Section 8.5.9. If the SIP Priority header is absent, and the P-CSCF has determined that the call attempt is an emergency call, the P-CSCF SHOULD add a SIP Priority header with the value of 'emergency' prior to forwarding the SIP message to the E-CSCF.

A SIP Priority header with a value of "emergency" SHOULD be added in all SIP requests related to PacketCable emergency calls, and as permitted within the [RFC 3261]. The header SHOULD be added by the UE or the P-CSCF and passed unscreened to the PSAP.

The following is an example of the Priority header:

```
Priority: emergency
```

Note: In case the P-CSCF decides to proceed with an emergency call that is not recognized as an emergency call by the UE as specified in the previous section, the P-CSCF MUST ignore the received Route header and use its provisioned E-CSCF route.

8.6.3 Handling of Network-Initiated De-Registration during Emergency Calls

As defined in [PKT 24.229], network-initiated de-registration may occur; it should, however, be handled carefully when an emergency call is attempted or in progress.

The P-CSCF MUST defer the processing of a network-initiated de-registration request for a Public Identity if it determines that an emergency call has been initiated or is currently in progress for the given Public Identity.

8.7 E-CSCF Requirements

8.7.1 Emergency Call Routing in the NENA i3 Architecture

8.7.1.1 Determining the Request-URI to forward the SIP request to

The E-CSCF MUST use the location reported inside the SIP message body to determine the correct PSAP to which to route the call, either through a static mapping or through the use of a mapping protocol.

Note: A candidate mapping protocol is the Location-to-Service Translation Protocol [ID LOST], which is being defined by the IETF ECRIT working group.

If the mapping returns a response indicating that no information is available (the mapping function is operable, but does not have data to determine the correct URI), the E-CSCF MUST follow the procedure in Section 8.7.2.

8.7.1.2 Processing the INVITE request after successful ECRIT mapping resolution

The E-CSCF MUST inspect the INVITE request for the presence of a P-Asserted-Identity header and drop it if it is present.

The E-CSCF MUST generate and insert a new P-Asserted-Identity header with a tel: URI of the actual caller's Telephone Number. The actual telephone number MUST always be provided, regardless of any call features (unlisted number, temporary or permanent caller-id blocking, etc.), in order to allow the PSAP to call the subscriber back.

The Request-URI MUST be changed to the URI determined by procedures in Section 8.7.1 above.

The SIP INVITE initiating an emergency call should be forwarded to the URI determined by procedures in Section 8.7.1 above.

8.7.2 Emergency Call Routing in the NENA i2 Architecture

8.7.2.1 Emergency Call Routing in NENA i2: determining ESRN and ESQK

The E-CSCF receives an INVITE request from the P-CSCF, which contains location as a PIDF-LO in the body of the INVITE, and the TN of the caller, either in the P-Asserted-Identity if present, in the P-Preferred-Identity if present, or, otherwise, in the From header.

The E-CSCF SHOULD query a [NENA i2] compliant VoIP Positioning Center using the "V5" interface, sending the location and TN of the caller as described in [NENA i2]. The VPC returns an ESRN, ESQK and Last Resort Option.

8.7.2.2 Processing the INVITE request in NENA i2

The E-CSCF MUST insert the ESQK as the TN in the INVITE as follows. If a P-Asserted-Identity header is present in the INVITE, then the E-CSCF MUST delete that P-Asserted-Identity header value, and replace it with a P-Asserted-Identity header value containing the ESQK as the TN. If the P-Asserted-Identity header is not present, and the P-Preferred-Identity header is present in the INVITE, then the E-CSCF MUST delete the P-Preferred-Identity header value, and replace it with a P-Asserted-Identity containing the ESQK as the TN. If neither a P-Asserted-Identity nor a P-Preferred-Identity is present, then the E-CSCF MUST create a P-Asserted-Identity containing the TN in the received From header.

The E-CSCF MUST then set the Request URI to the ESRN. The E-CSCF MUST add the following SIP headers: a History header, a Via header, and a Record-Route header.

The E-CSCF MUST then route the INVITE to the SG or MGC.

8.7.2.3 Dealing with errors

If a call is returned to the E-CSCF by the ESGW network, the E-CSCF MUST restore the header used to carry the ESQK to the value it received (that is, the TN of the caller), change the Request-URI to the Last Routing Option (LRO), and forward the call to a PSTN gateway.

If a call is returned by the PSTN gateway, the E-CSCF MUST return a 600 Busy Everywhere.

8.7.2.4 Processing a BYE

When the E-CSCF receives the BYE transaction, it MUST notify the VPC, using the V5 interface, to release the ESQK.

The E-CSCF MUST then forward the BYE towards the P-CSCF.

8.7.3 Emergency Call Routing in NENA Pre-i2

The call is routed by the dedicated E911 PSTN network selective router based on the telephone number of the caller, and the location associated with the telephone number in the selective router.

8.7.3.1 INVITE Request Processing

E-CSCF MUST insert a P-Asserted-Identity in the outgoing INVITE. If a P-Asserted-Identity is present in the received INVITE, then this P-Asserted-Identity MUST be inserted in the outgoing INVITE. If a P-Asserted-Identity is not present, then if a P-Preferred-Identity is present in the received INVITE, then the E-CSCF MUST create a P-Asserted-Identity derived from the P-Preferred-Identity. If neither a P-Asserted Identity nor a P-Preferred-Identity is present, then the E-CSCF MUST create a P-Asserted-Identity derived from the From: header.

If the P-Asserted-Identity header is not in a telephone number format, the call can still be routed, and is default routed by the selective router.

No location information is required, but if it is provided, it MUST be passed in the INVITE towards the MGC.

Since the location based routing is done in the selective router, the E-CSCF MUST route the call to the MGC and the Selective Router.

8.7.3.2 Additional message processing

No additional message processing is required.

8.7.4 Emergency Call Routing in the NENA i1 Architecture

The call is routed to the PSTN number associated with the PSAP. This solution does not make use of the dedicated TDM selective router.

8.7.4.1 Processing Invite Request

In these scenarios, no location information is required, but if the location is received in the INVITE, the E-CSCF MUST insert it in the INVITE to be sent towards the MGC, SG, and MG.

The E-CSCF MUST convert the SIP-Request-URI to the E.164 PSTN number of the PSAP. The method by which this is done is beyond the scope of this specification.

The E-CSCF MUST then route the INVITE to the SG/MGC/MG.

8.7.4.2 Processing additional messages

No additional message processing is required.

8.8 MGC, SG and MG Requirements

This section identifies requirements on the MGC, SG, and MG for emergency calls. These components interface with the PSTN and the selective router in the PacketCable architecture. In order to facilitate the PSTN connections, compliance with Telcordia PSTN and selective router interface documents is required.

8.8.1 Emergency Call Routing in NENA i3

The MGC, SG, and MG are not used in [NENA i3] because the PSAP is directly connected to the PacketCable network via SIP.

8.8.2 Emergency Call Routing in NENA i2

The MGC requirements for emergency call routing in NENA i2 are specified in [CMSS1.5]. The MG requirements for emergency call signaling in NENA i2 are specified in [TGCP1.5].

8.8.3 Emergency Call Routing in NENA Pre-i2

The MGC requirements for emergency call routing in NENA Pre-i2 are specified in [CMSS1.5]. The MG requirements for emergency call signaling in NENA Pre-i2 are specified in [TGCP1.5].

8.8.4 Emergency Call Routing in the NENA i1 Architecture

The call is treated as a normal PSTN call. No special requirements are needed on the MGC, SG, and MG. Network Hold and Network Ringback are not supported.

Annex A Digit Map Syntax (Normative)

The following Augmented Backus-Naur Form describes the digit map syntax for UEs:

```
; First, some preliminaries

Alphanum            = ALPHA / DIGIT
Identifier          = ALPHA *( [ "_" / "-" ] 1*Alphanum )
Number              = 1*DIGIT
Fraction            = ( Number ["." [Number]] / "." Number )

CommentText         = VCHAR / WSP
Comment             = "//" *CommentText CRLF   ; end-of-line comments like C++
LineBreak           = 1*(*WSP (CRLF / Comment))

LWSC                = *LineBreak 1*WSP ; Linear WhiteSpace w/ optional Comment
SWSC                = [LWSC]                   ; Separating Whitespace
                                                     ; w/ optional Comment

COMMA               = SWSC "," SWSC
SEMI                = SWSC ";" SWSC
COLON               = SWSC ":" SWSC
EQUAL               = SWSC "=" SWSC
LPAREN              = SWSC "(" SWSC
RPAREN              = SWSC ")" SWSC

SQUOTE              =  "'"                     ; Single Quote

NonDQuote           = %x21 / %x23-7E           ; VCHAR less DQUOTE
NonSQuote           = %x21-%x26 / %x28-7E      ; VCHAR less SQUOTE

MapNameDef          = Identifier               ; distinguished semantically
MapNameRef          = Identifier               ; distinguished semantically
SymbolNameDef       = Identifier               ; distinguished semantically
SymbolNameRef       = Identifier               ; distinguished semantically

; The major substance follows

DigitMapPackage     = *(TimerDef / SymbolDef / MapDef / LineBreak)

TimerDef            = "Timer" 1*WSP Timer EQUAL Fraction LineBreak
Timer               =      StandaloneTimer / PrefixTimer
StandaloneTimer     = "T" / "S" / "L"
PrefixTimer         = "Z"                      ; modifies the following key

SymbolDef           = SymbolNameDef EQUAL Constant *( SWSC Constant ) LineBreak
Constant            = DQUOTE 1*NonDQuote DQUOTE
                           / SQUOTE 1*NonSQuote SQUOTE

MapDef              = "Map" 1*WSP MapNameDef EQUAL 1*( LWSC Rule LineBreak)

Rule                = Pattern COLON Action *(SEMI Action)

Pattern             = DQUOTE PatternBody DQUOTE
PatternBody         = 1*( Target [Count])

Target              = [ PrefixTImer ] Key      ; Keypress with optional
                                                     ; long duration
                           / StandaloneTimer         ; Delay triggering match
                           / SubPattern
                           / Noise                   ; ignored, just for readability

SubPattern          = "(" PatternBody ")"      ; nested patterns for repetition
                                                     ; and referencing
```

```
                           / "(" "=" Reference ")"    ; ref to map or symbol

Key                        =      DIGIT               ; dialled digit
                           / "X"                      ; any numeric dialled digit
                           / "*" / "#"                ; special dialled symbols
                           / "A" / "B" / "C" / "D"    ; extra number pad keys
                           / "[" ["^"] DIGIT "-" DIGIT "]"
                                                      ; any single dialled digit in
range
                                                      ; or any dialled char not in rang

Count               = "{" [Number "-"] Number "}"
                                                      ; repeat no fewer than 1st #,
                                                      ; no more than last #
Noise               = "-" / "." / WSP
Reference           = MapNameRef / SymbolNameRef

Action              = Verb [LPAREN Parameter *( COMMA Parameter ) RPAREN]
Verb                = "RETURN" / "USEMAP" / "ACR-ACTIVATE" / "ACR-DEACTIVATE" /
"AC-ACTIVATE" / "MAKE-CALL" / "AR-ACTIVATE" / "CID-SUPPRESS" / "CID-DELIVER" /
"COT-ACTIVATE" / "DA-MAINT" / "DND-PROGRAM" / "HOLD-ACTIVATE" / "SCF-PROGRAM" /
"SB-MAINT" / "SD-PROGRAM" / "SPP-PROGRAM" / "EMERGENCY-CALL" / Identifier
Parameter           = StringParam / MapParam
StringParam         = ParamPiece *( [SWSC] ParamPiece )
ParamPiece          = Constant
                           / "=" SymbolNameRef
                           / "#" Number
MapParam            = "=" MapNameRef
```

Appendix I Acknowledgements

PacketCable wishes to recognize the following individuals for their significant involvement and contributions to this specification (ordered alphabetically by company name and individual's first names in each company):

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