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IPCablecom2 residential SIP telephony: Feature definition

Recommendation ITU-T J.460.0

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Summary

Recommendation ITU-T J.460.0 describes the IPCablecom2 residential SIP telephony feature definitions and underlying assumptions. The IPCablecom2 residential SIP telephony features include many of the features commonly available to residential subscribers of public switched telephone network services. Examples include caller ID, call waiting, speed dialing, and emergency calling capabilities. This Recommendation also describes the expectations for the devices that can be used for IPCablecom2 residential SIP telephony.

Source

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FOREWORD

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CONTENTS

		Page
1	Scope	1
2	References.....	1
	2.1 Informative References	1
	2.2 Reference Acquisition	2
3	Terms and definitions	2
4	Abbreviations and acronyms	3
5	IPCablecom2 Residential SIP Telephony Scope.....	4
	5.1 Overview	5
	5.2 Residential SIP Telephony Feature Groupings	6
6	IPCablecom2 Residential SIP Telephony Feature Definitions.....	6
	6.1 Basic Calling Capabilities	7
	Appendix I – Requirements for Region A (Primarily Europe).....	12
	Appendix II – Requirements for Region B (Primarily North America)	13
	II.1 Scope	13
	II.2 References	13
	II.3 Terms and definitions	15
	II.4 Abbreviations and acronyms	15
	II.5 IPCablecom2 Residential SIP Telephony Scope.....	15
	II.6 IPCablecom2 Residential SIP Telephony Feature Definitions	15
	Appendix III – Region C of Residential SIP Telephony Feature Definition	40
	III.1 Scope	40
	III.2 References	40
	III.3 Terms and definitions	41
	III.4 Abbreviations and acronyms	41
	III.5 IPCablecom2 Residential SIP Telephony Scope.....	41
	III.6 IPCablecom2 Residential SIP Telephony Feature Definitions	41
	Appendix IV – User Interface Tones and Signals.....	42
	Appendix V – Vertical Service Codes	46

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IPCablecom2 residential SIP telephony: Feature definition

1 Scope

The IPCablecom2 residential SIP telephony effort is intended to replicate many of the residential features currently available on the public switched telephone network (PSTN), except that the features operate using an IPCablecom2-based IP (Internet Protocol) network using session initiation protocol for voice over IP (SIP VoIP) sessions. The IPCablecom2 residential SIP telephony features described in this Recommendation are intended to form the basis for specification requirements that define the behaviour of SIP clients that interact with the IPCablecom2 network and the corresponding network behaviour.

This Recommendation describes the behaviour of residential telephony features using a standard RST Client (called RST User Equipment, or RST UE, or simply UE in this Recommendation) connected to the IPCablecom2 network.

Prior IPCablecom efforts have defined residential telephony capabilities that operate on an IPCablecom IP network. IPCablecom defined an MGCP-based (Media Gateway Control Protocol) architecture in which a centralized call management server (CMS) controls the telephony services provided by multimedia terminal adapters (MTAs).

The IPCablecom2 residential SIP telephony specification defines how UEs can support residential telephony capabilities similar to that of IPCablecom. This allows for the deployment of new types of clients and services while maintaining continuity with existing residential telephony services.

This Recommendation is the overview document describing basic telephony features to be supported in SIP. The Recommendations that specify the implementation of these features are given below.

- J.460.1 IPCablecom2 residential SIP telephony feature specification
- J.460.2 IPCablecom2 residential SIP telephony E-DVA specification
- J.460.3 IPCablecom2 residential SIP telephony accounting specification
- J.460.4 IPCablecom2 residential SIP telephony E-UE provisioning specification.

It is an important objective of this work that interoperability between IPCablecom 2.0 and 3GPP IMS is provided. IPCablecom 2.0 is based upon 3GPP IMS, but includes additional functionality necessary to meet the requirements of cable operators. Recognizing developing converged solutions for wireless, wireline, and cable, it is expected that further development of IPCablecom 2.0 will continue to monitor and contribute to IMS developments in 3GPP, with the aim of alignment of 3GPP IMS and IPCablecom 2.0.

NOTE – The structure and content of this Recommendation have been organized for ease of use by those familiar with the original source material; as such, the usual style of ITU-T recommendations has not been applied.

2 References

2.1 Informative References

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- [ARCH-FRM TR] Recommendation ITU-T J.360 (2006), *IPCablecom2 architecture framework*.



- [CMSS1.5] Recommendation ITU-T J.178 (2005), *IPCablecom CMS to CMS signalling*.
- [CODEC1.5] Recommendation ITU-T J.161 (2007), *Audio and video codec requirements and usage for the provision of bidirectional audio services over cable television networks using cable modems*.
- [CODEC-MEDIA] Recommendation ITU-T J.361 (2006), *IPCablecom2 Codec and Media*.
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- [ITU-T V.152] Recommendation ITU-T V.152 (2005), *Procedures for supporting voice-band data over IP networks*.
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- [ITU-T V.18] Recommendation ITU-T V.18 (2000), *Operational and interworking requirements for DCEs operating in the text telephone mode*.
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- [ITU-T V.8] Recommendation ITU-T V.8 (2000), *Procedures for starting sessions of data transmission over the public switched telephone network*.
- [NCS1.5] Recommendation ITU-T J.162 (2005), *Network call signalling protocol for the delivery of time-critical services over cable television networks using cable modems*.
- [OB 781] Operational Bulletin No. 781 (1.II.2003) and Annexed List: *Various tones used in national networks (According to ITU-T Recommendation E.180) (03/1998) (Position on 1 February 2003)*.
- [RFC 2833] IETF RFC 2833 (2000), *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*.
- [RFC 3261] IETF RFC 3261 (2002), *SIP: Session Initiation Protocol*.

2.2 Reference Acquisition

- Internet Engineering Task Force (IETF) Secretariat c/o Corporation for National Research Initiatives, 1895 Preston White Drive, Suite 100, Reston, VA 20191-5434, Phone 703-620-8990, Fax 703-620-9071, Internet: <http://www.ietf.org/>

3 Terms and definitions

This Recommendation uses the following terms and definitions:

- 3.1 jitter:** Variability in the delay of a stream of incoming packets making up a flow such as a voice communication.
- 3.2 latency:** The time taken for a signal element to pass through a device or circuit.
- 3.3 media gateway:** Devices bridging between the IPCablecom IP Voice Communication network and the PSTN. A Media Gateway provides the bearer circuit interfaces to the PSTN and transcodes the media stream.

3.4 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.

3.5 on hook: The idle state of a traditional telephone, while no call is in progress and the telephone handset is sitting in its cradle.

3.6 SIP client: The functional element used by subscribers to attach to the IPCablecom2 network.

3.7 trunk: An analog or digital connection from a circuit switch that carries user media content and may carry voice signaling (MF, R2, etc.).

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

3WC	Three Way Calling
AC	Auto Callback or Alternating Current
ACR	Anonymous Call Rejection
AR	Auto Recall
AS	Application Server
BCT	Blind Call Transfer
BLV	Busy Line Verification
CCT	Consultative Call Transfer
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
CNAM	Calling Name
CND	Calling Number Delivery
COT	Customer Originated Trace
CPE	Customer Premises Equipment
CSCF	Call Session Control Function
CT	Call Transfer
DC	Direct Current
DN	Directory Number
DND	Do Not Disturb
DTMF	Dual Tone Multi Frequency
DVA	Digital Voice Adapter
E-DVA	Embedded Digital Voice Adapter
EI	Emergency Interrupt

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E-MTA	Embedded Multimedia Terminal Adapter
FSK	Frequency Shift Key
IMS	IP Multimedia Subsystem
IP	Internet Protocol
LCD	Liquid Crystal Display
MGC	Media Gateway Controller
MTA	Multimedia Terminal Adapter
NAT	Network Address Translation
NCS	Network Call Signaling
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RACF	Remote Activation of Call Forwarding
ROH	Receiver-Off-Hook
RST	Residential SIP Telephony
RTP	Real-time Transport Protocol
SCA	Selective Call Acceptance
SCF	Selective Call Forwarding
SCR	Selective Call Rejection
SIP	Session Initiation Protocol
SLE	Screening List Editing
SPP	Subscriber Programmable PIN
TDD	Telecommunications Device for the Deaf
UE	User Equipment
URI	Uniform Resource Identifier
URL	Uniform Resource Locator
VMWI	Visual Message Waiting Indicator
VoIP	Voice over Internet Protocol

5 IPCablecom2 Residential SIP Telephony Scope

The IPCablecom2 Residential SIP Telephony Feature Definition describes a grouping of telephony call features and capabilities that approximate the services available to most PSTN residential voice subscribers. The IPCablecom2 Residential SIP Telephony is an application built upon the IPCablecom2 architecture described in [ARCH-FRM TR].

The IPCablecom2 architecture framework Recommendation and related specifications specify the system components, the various functional groupings and the network interfaces necessary for delivery of services via an IPCablecom2 network. The architecture (which is based on IMS) leverages the use of triggers within the core network to determine when application level processing is required.

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

This clause provides:

- An overview of the IPCom2 Residential SIP Telephony feature capabilities.
- A description of the main feature groupings.

5.1 Overview

This Recommendation defines the IPCom2 Residential SIP Telephony features from a user's point of view, without delving into many of the details of the network elements in the IPCom2 network.

Figure 1 illustrates the context for this Recommendation, and the interfaces that are within scope for this Recommendation.

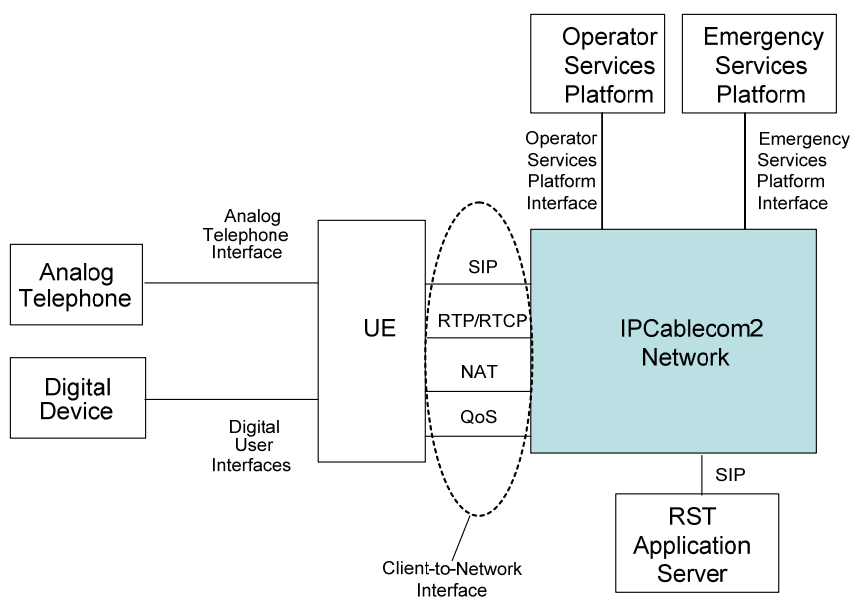


Figure 1 – IPCom2 Residential SIP Telephony Feature Definition Context

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

UE: the functional element subscribers use to attach to the IPCom2 network. UEs may take any number of forms, including clients embedded into cable modems, clients embedded into set-top boxes, clients embedded into SIP Phones, and software clients on PCs or laptops. UEs support several interfaces to the IPCom2 network:

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

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

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

RST Application Server: The Residential SIP Telephony Application Server is a logical network component that implements some aspects of RST features. The IPCom2 architecture provides a framework for triggering SIP messages to the application server based on the service profile of a user.

Operator Services Platform: Some Residential SIP Telephony features require the intervention of an operator. This Recommendation describes the interactions between the IPCom2 network and the Operator Services Platform.

Emergency Services Platform: Residential SIP Telephony features require specialized capabilities supporting emergency calling. This Recommendation describes the interactions between the IPCom2 network and the Emergency Services Platform.

5.2 Residential SIP Telephony Feature Groupings

The Residential SIP Telephony capabilities are divided into feature groups for easy reference. The feature groups are defined below:

- **Basic Calling Capabilities:** including call origination, call termination, addressing, quality of service, and basic call progress indicators.
- **Caller ID:** including all features associated with delivery and blocking of a caller's name and number.
- **Call Forwarding:** including all supported variants of the call forwarding feature.
- **Call Blocking:** including support for blocking specific categories of outbound and inbound calls.
- **Multi-Party Features:** including call waiting, hold, transfer, and three-way calling.
- **Miscellaneous Features:** including distinctive alerting, speed dialing, and customer-originated call trace features.
- **Emergency Services:** including emergency calling and location identification capabilities.
- **Operator Services:** including busy-line verification and emergency operator interruption capabilities.
- **Lifeline Capabilities:** including capabilities for providing service during AC power outages.

6 IPCom2 Residential SIP Telephony Feature Definitions

This clause contains detail on each of the call features that may be offered as part of IPCom2 Residential SIP Telephony. Service definitions are described for the simple case of a single UE emulating the legacy PSTN residential voice service, and for the case of how similar goals can be met in an IPCom2 SIP-enabled environment, where multiple UEs can be a part of the same Residential SIP Telephony subscription.

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

- **IPCablecom2 Residential SIP Telephony Feature Specification**
This specification defines specific requirements on the UE and Application Servers to implement Residential SIP Telephony features. Some of the features are completely implemented on the UE; some are completely implemented on an Application server, while some others are implemented on both UE and Application Server. The MGC requirements for support of Residential SIP Telephony features are specified in [CMSS1.5].
- **IPCablecom2 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.
- **IPCablecom2 Residential SIP Telephony E-DVA Specification**
This specification defines the embedded Digital Voice Adaptor (E-DVA) requirements for the analog interface and for powering of the E-DVA. A DVA is a UE that supports an analog phone interface. An embedded DVA is a DVA integrated with a cable modem.

6.1 Basic Calling Capabilities

6.1.1 UEs and Public Identities

One of the major differences between a traditional telephone network and a SIP Telephony network lies in the relationships between users, public identities, UEs, and devices. A user may have multiple devices, each of which may host multiple UEs, and each UE is registered to one or more public identities. A public identity can be a telephone number or it can be an alphanumeric identifier that makes sense in the context of a SIP Telephony service. Each public identity is generally associated with a user.

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

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

6.1.2 Basic Call for Multiple UEs per Public Identity

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

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

NOTE – The first phase of RST specifications assumes that there is a single UE behind a given public identity.

6.1.3 Addressing

6.1.3.1 Dial String Entry

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

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

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

6.1.3.2 Initiating a Call or Feature via Dial String

With traditional analog telephone devices, it is customary for features to be invoked by entering a dial string that denotes the feature requested as well as data pertinent to that feature. Features are often, but not always, denoted by star-codes (dial strings that begin with "*", see Appendix V) whereas dial strings consisting exclusively of digits usually denote a request for call initiation. In IP-Cablecom2 calls are initiated by a UE, while features may be performed by the UE, an element in the network, or a combination of the two.

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

6.1.3.3 Creating a Routable Address from a Dial String

IP-Cablecom2 Residential SIP Telephony is intended to serve as an alternative to conventional PSTN service. This requires that it support the same kinds of dial strings used in the PSTN, and that it support calling addresses in the PSTN.

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

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

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

6.1.4 Call Progress Indications

Call progress indications are signals that inform the user of call progression status. Call progress indications are applied by the originating UE to inform the user of the status of an attempted call, usually in the form of tones played to the calling user through the earpiece of the telephone handset (in the case of a traditional telephone user interface). Call progress indications can also be used to notify a user that a feature is activated or deactivated. An example would be the transmission of a stutter dial tone. Call progress tones defined for IP-Cablecom2 are listed in Appendix IV.

6.1.5 Announcements

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

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

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

6.1.6 Permanent Sequence and Lockout

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

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

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

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

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

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

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

6.1.7 No Answer Timeout

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

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

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

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

6.1.8 UE Status

When a UE knows it has become available or unavailable to receive calls, it signals its availability or unavailability to the IP-Cablecom2 network by registering (or de-registering) with a configured public identity.

If a call is placed to a public identity for which there are no registered UEs, then the public identity is considered unreachable, not busy. An unreachable UE has similar semantics to a mobile phone that is out of range of a mobile phone provider's network. This can be an important semantic distinction for features that are defined in terms of "busy" or "no answer" conditions, such as Call Forwarding Busy Line (CFBL), CFDA, and call forwarding to voice mail. In the case of the CFBL feature, an unreachable UE does not trigger CFBL forwarding, but may trigger call forwarding to voice mail or CFDA.

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

6.1.9 Quality of Service

IPCablecom2 Residential SIP Telephony utilizes the Quality of Service (QoS) capabilities of the IPCablecom2 architecture as defined in [ARCH-FRM TR]. In general, if QoS resources cannot be allocated for a call, operator policy dictates whether the call is permitted to proceed or not. One operator may not permit calls for which QoS cannot be allocated to proceed while another operator may permit such calls. In the case of an emergency services call, or an operator services emergency interruption, the call may be attempted even if QoS resources cannot be allocated for the call, per operator policy. Also, per operator policy, emergency calls may be granted higher-priority QoS than other, non-emergency calls.