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IPCablecom2 residential SIP telephony: Feature specification

Recommendation ITU-T J.460.1

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Recommendation ITU-T J.460.1

IPCablecom2 residential SIP telephony: Feature specification

Summary

Recommendation ITU-T J.460.1 specifies an implementation of common residential telephony features in an IP-Cablecom2 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. This Recommendation defines the Layer 3 and higher network requirements and signaling to enable User Equipment (UE) to establish services to emulate historical analog telephony services and features.

Source

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Recommendation ITU-T J.460.1

IPCablecom2 residential SIP telephony: Feature specification

1 Scope

This Recommendation specifies an implementation of common residential telephony features in an IPCablecom2 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 IPCablecom2 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 IPCablecom2 architecture is based. Many telecom equipment and software suppliers are focusing their investments in converged network solutions on the IMS architecture, and the IPCablecom2 architecture leverages the IMS architecture, which should yield more rapid implementation timeframes.

This initial Residential SIP Telephony Feature Specification defines the Layer 3 and higher network requirements and signaling to enable User Equipment (UE) to establish services to emulate historical analog telephony services and features. This release of the Recommendation supports services described in Appendix II of [RST TR]. Future releases will support additional services that are described in Appendices I and III of [RST TR] as they are developed.

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 Normative References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

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- [RST TR] Recommendation ITU-T J.460.0, *IP-Cablecom2 – Residential SIP Telephony: Feature: Definition*.
- [TGCP1.5] Recommendation ITU-T J.171.1: *IP-Cablecom trunking gateway control protocol (TGCP): Profile 1*.

2.3 Reference Acquisition

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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/lid-abstracts.txt>.
- National Emergency Number Association (NENA), http://www.nena.org/9-1-1TechStandards/tech_info_docs.htm, <http://www.nena.org/9-1-1TechStandards/voip.htm>
- Telcordia Technologies, <http://www.telcordia.com/>

3 Terms and Definitions

This Recommendation defines the following terms:

- 3.1 active call/active session:** A call state where the called party has answered the call and two-way media is being exchanged.
- 3.2 automatic location identification (ALI):** The database that maps telephone number to location in the current 9-1-1 system.
- 3.3 automatic number identification (ANI):** The mechanism used to determine the telephone number of the caller.
- 3.4 callback:** When a PSAP initiates a new call back to a caller. This is not the same as ringback. See below.
- 3.5 CNAM:** Calling Name.
- 3.6 configuration server (Config. Server):** The logical network element responsible for UE provisioning, configuration and management.
- 3.7 dynamic host configuration protocol (DHCP):** 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.

- 3.8 emergency services gateway (ESGW):** A device which bridges the VoIP network to the Selective Router. In IPCablecom2 this is a function of the MGC, SG and MG.
- 3.9 emergency services query key (ESQK):** 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.
- 3.10 emergency services routing number (ESRN):** 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.
- 3.11 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.
- 3.12 i3:** The third and final migration phase planned by NENA, a more long-term approach to address the needs of having IP-enabled emergency centres.
- 3.13 integrated services digital network (ISDN):** User Part (part of the SS7 signaling stack).
- 3.14 internet engineering task force (IETF):** The standards body for the Internet.
- 3.15 media gateway:** Devices bridging between the IPCablecom2 IP Voice Communication network and the PSTN. A Media Gateway provides the bearer circuit interfaces to the PSTN and transcodes the media stream.
- 3.16 national emergency number association (NENA):** The primary standards body for the 9-1-1 community.
- 3.17 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.
- 3.18 on hook:** The idle state of a traditional telephone, while no call is in progress and the telephone handset is sitting in its cradle.
- 3.19 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.
- 3.20 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.
- 3.21 SIP Client:** The functional element subscribers use to attach to the IPCablecom2 network.
- 3.22 (ENUM) Telephone Number Mapping:** In [RFC 3761], the E.164 to Uniform Resource Identifiers (URI) Dynamic Delegation Discover System (DDDS) Application (ENUM).
- 3.23 trunk:** An analog or digital connection from a circuit switch that carries user media content and may carry voice signaling (MF, R2, etc.).
- 3.24 VoIP Positioning Centre:** 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, acronyms and conventions

4.1 Abbreviations and Acronyms

This Recommendation uses the following abbreviations:

3WC	Three Way Calling
AC	Auto Callback or Alternating Current

ACR	Anonymous Call Rejection
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 and Suppression
CND	Calling Number Delivery
COT	Customer Originated Trace
CPE	Customer Premises Equipment
CT	Call Transfer
CWT	Call Waiting
DND	Do Not Disturb
DTMF	Dual Tone Multi Frequency
DVA	Digital Voice Adapter
E-CSCF	Emergency-Call Session Control Function
ENUM	Telephone Number Mapping
GRUU	Globally Routable User-Agent URI
HSS	Home Subscriber Server
ICE	Interactive Connectivity Establishment
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ISUP	Integrated Services Digital Network (ISDN) User Part
IVR	Interactive Voice Responder
MG	Media Gateway
MGC	Media Gateway Controller
NAT	Network Address Translation
NCS	Network Call Signalling
OCB	Outbound Call Blocking
P-CSCF	Proxy – Call Session Control Function
PIDF-LO	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	Signalling Gateway
SIP	Session Initiation Protocol, VoIP signaling protocol, defined by [RFC 3261]
SLE	Screening List Editing
SPP	Subscriber Programmable PIN
SS7	Signalling System #7
STUN	Simple Traversal of UDP through NATs
TCAP	Transaction Capabilities Application Part (part of the SS7 signalling 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 Centre
VSC	Vertical Service Code
XCAP	XML Configuration Access Protocol

4.2 Conventions

Throughout this Recommendation, 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 Recommendation.
"MUST NOT"	This phrase means that the item is an absolute prohibition of this Recommendation.
"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 behaviour is acceptable or even useful, but the full implications should be understood and the case carefully weighed before implementing any behaviour 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.

5 Overview

This Residential SIP Telephony Feature Specification is an application Recommendation that relies upon the IP-Cablecom2 architecture as described in [ARCH-FRM TR] and related IP-Cablecom2 Recommendations. The Residential SIP Telephony Recommendations are embodied in a separate standalone release. This Recommendation specifies how residential telephony features must be implemented in IP-Cablecom2 networks.

5.1 IP-Cablecom2 Design Considerations for Residential SIP Telephony

The following design principles have been used in the creation of this Recommendation:

- *Minimal state awareness in network elements:*
This Recommendation 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 Recommendation 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 Recommendation avoids mandating changes to the IMS core network when it is reasonable to do so. This Recommendation primarily defines additional requirements for User Equipment, Application Servers, and PSTN Interconnect elements such as Media Gateway Controllers.
- *Transparency of end-user behaviour 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 behaviour and feature interaction of some telephony features may differ from that described within Telcordia LSSGR specifications. In most cases, the user behaviour described within the referenced LSSGR specifications has been reproduced.

5.2 Residential SIP Telephony Specific Applications

This Recommendation 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 Recommendation), which is in turn connected to an IP-Cablecom2 network.

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

This Recommendation 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

For a description of the IPCablecom2 architecture and Functional Components, please refer to [ARCH-FRM TR].

This Recommendation 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 Recommendation, every feature that requires an Application Server is associated with a distinct one. For example, the Call Transfer clause defines requirements on the Call Transfer Application Server (CT AS); the Outbound Call Blocking clause 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 Recommendation (this Recommendation) is part of the RST suite of Recommendations which also consists of the following:

- *IPCablecom2 Residential SIP Telephony Accounting Specification*
The purpose of this Recommendation 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 Recommendation 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 an IPCablecom2 DVA.

6 Normative Reference Application

See clause I.6.

7 Residential SIP telephony feature requirements

7.1 Basic Calling Capabilities

All call flows in this Recommendation 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 Recommendation.

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 clause is for North American digit maps, the IPCablecom2 digit map syntax defined below can represent any country digit map.



IPCablecom2 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 D 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:** The allowable time between digits if the short interdigit timer has not been indicated in the digit map.
- **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 SymbolDef's 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 parentheses '(' 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 behaviour 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, followed by a 'Y'. 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 returned 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" #1y =domain =dialstring )
    "0([2-9]x{0-15})S" : 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( "01" #1 )
    "01([2-9]x{1-14})#" : RETURN( "01" #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 SEMI "cic=+1" #1 )
                                      // (3 digit CIC dialed by user)
    "101(x{4})(=PhoneNumbers)" : RETURN( #2v SEMI "cic=+1" #1 )
                                   // (4 digit CIC dialed by user)

MAP ImmediateVSCs = // Matches and executes immediate VSCs. Returns nothing
    "*57" // (COT-ACTIVATE) : COT-ACTIVATE // Cannot use MAKE-CALL for this.
    "*63" // (SCF) : MAKE-CALL ( "sip:" #0" =domain =dialstring)
    "*66" // (AC-ACTIVATE) : AC-ACTIVATE // Auto Callback
    "*86" // (AC-DEACTIVATE) : AC-DEACTIVATE // Auto Callback
    "*69" // (AR-ACTIVATE) : AR-ACTIVATE // Auto Recall
    "*89" // (AR-DEACTIVATE) : AR-DEACTIVATE // 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-SUPPRESS : CID-SUPPRESS
    "*82" // CIDS-DELIVER : CID-DELIVER
    "*70" // (CCS) (Really Toggle-Call-Waiting)
        : CW-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 IPCom2 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. The specific VSCs used are a national matter.

Table 1 provides the defined local action identifiers for the services defined in this Recommendation.

Table 1 – Digit Map Actions

Service	Action	UE Action ID
Anonymous Call Rejection	Activation	ACR-ACTIVATE
	Deactivation	ACR-DEACTIVATE
Automatic Callback	Activation	AC-ACTIVATE
	Deactivation	AC-DEACTIVATE
Basic Call	Activation	MAKE-CALL
Automatic Recall	Activation	AR-ACTIVATE
	Deactivation	AR-DEACTIVATE
Caller ID Per-Call Blocking	Suppress	CID-SUPPRESS
Caller ID Per-Call Delivery	Deliver	CID-DELIVER
Customer Originated Trace	Activation	COT-ACTIVATE
Call Waiting	Toggle	CW-TOGGLE
Call Forward Variable	Programming	CFV-PROGRAM
	Deactivation	CFV-DEACTIVATE

Table 1 – Digit Map Actions

Service	Action	UE Action ID
Distinctive Alerting	List Maintenance	DA-MAINT
Do Not Disturb	Programming	DND-PROGRAM
Hold	Activation	HOLD-ACTIVATE
Selective Call Forwarding	Programming	SCF-PROGRAM
Solicitor Blocking	List Maintenance	SB-MAINT
Speed Dial	Programming	SD-PROGRAM
Subscriber Programmable PIN	Programming	SPP-PROGRAM
Emergency Call	Activation	EMERGENCY-CALL

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 behaviour, 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 the Configuration Server.

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 clause apply in the absence of other features.

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].

- In the IP packets carrying the media sent from the UE towards the network, set the DSCP field to the provisioned Non-Emergency media DSCP Value defined in Table 2.

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 3.

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 2.

Table 2 – 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

If the Calling UE is configured to require that network resources are reserved before continuing with the session ([PKT 24.229]), then the Calling UE MUST NOT deliver media to the user until resource reservation is complete. If the alert-info header is included in an 18X provisional response from the called party and resource reservation is required but not yet completed, then the Calling UE MUST delay the application of the alert-info header until an updated offer SDP is received showing that the remote resources are now reserved.

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 clause 7.1.6.7, respond with an ACK and terminate the session.

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

7.1.4.2 Receiving a Call

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

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

- In the IP packets carrying the media sent from the UE towards the network, set the DSCP field to the provisioned Non-Emergency media DSCP Value defined in Table 2.

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 3.

When a UE receives an incoming INVITE, and sends a progress message according to [PKT 24.229], it MUST alert the user unless resource reservation is in progress at either the originating or terminating UE. If the UE is configured to require that network resources are reserved before continuing with the session ([PKT 24.229]), and this reservation is not complete, then the Called UE MUST NOT alert the user until it has received an updated SDP from the calling party showing that the remote resources are now reserved.

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 clause I.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 3, '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 clause 7.1.7. After the Origination Mode Dial Tone timer defined in Table 3 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 3) for a time period 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 3 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 3. 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 3). 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 clause 7.1.7, or in termination mode with appropriate (ring, caller ID, etc.) signaling.

7.1.4.5 Feature Data

Table 3 summarizes the feature data elements defined to support implementation of the basic call feature.

Table 3 – Basic Call Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	Config. Server Requirement
Digit map	Digit Map	Volatile in UE	Per public user ID	Config. Server	Config. Server	UE	Mandatory
SDP parameters and value (Note)	Character string	Volatile in UE	Per public user ID	Config. Server	Config. Server	UE	Mandatory
BYE Delay	Integer (seconds)	Volatile in UE	Per network	Config. Server	Config. Server	UE	Mandatory
Origination Mode Dial Tone Timer	Integer (seconds)	Volatile in UE	Per network	Config. Server	Config. Server	UE	Mandatory
Termination Mode Off-Hook Error Signal	Tone	Volatile in UE	Per network	Config. Server	Config. Server	UE	Mandatory
Termination Mode Error Signal Timer	Integer (seconds)	Volatile in UE	Per network	Config. Server	Config. Server	UE	Mandatory

Table 3 – Basic Call Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	Config. Server Requirement
Permanent Sequence Tone 1	Tone	Volatile in UE, Non-volatile in Config. Server	Per network	Config. Server	Config. Server	UE	Mandatory
Permanent Sequence Timer 1	Integer (seconds)	Volatile in UE	Per network	Config. Server	Config. Server	UE	Mandatory
Permanent Sequence Tone 2	Tone	Volatile in UE	Per network	Config. Server	Config. Server	UE	Mandatory
Permanent Sequence Timer 2	Integer (seconds)	Volatile in UE	Per network	Config. Server	Config. Server	UE	Mandatory
Permanent Sequence Tone 3	Tone	Volatile in UE	Per network	Config. Server	Config. Server	UE	Mandatory
Permanent Sequence Timer 3	Integer (seconds)	Volatile in UE	Per network	Config. Server	Config. Server	UE	Mandatory
Lockout Reset Timer	Integer (seconds)	Volatile in UE	Per network	Config. Server	Config. Server	UE	Mandatory
Non-Emergency media DSCP Value	Integer (range of 0..63)	Volatile in UE	Per Network	Config. Server	Config. Server	UE	Mandatory

NOTE – 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 clause for feature interactions with basic calls.

7.1.4.7 Call Flows

The call flows in this clause 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 1 illustrates a call from one UE to another UE where the call is answered.

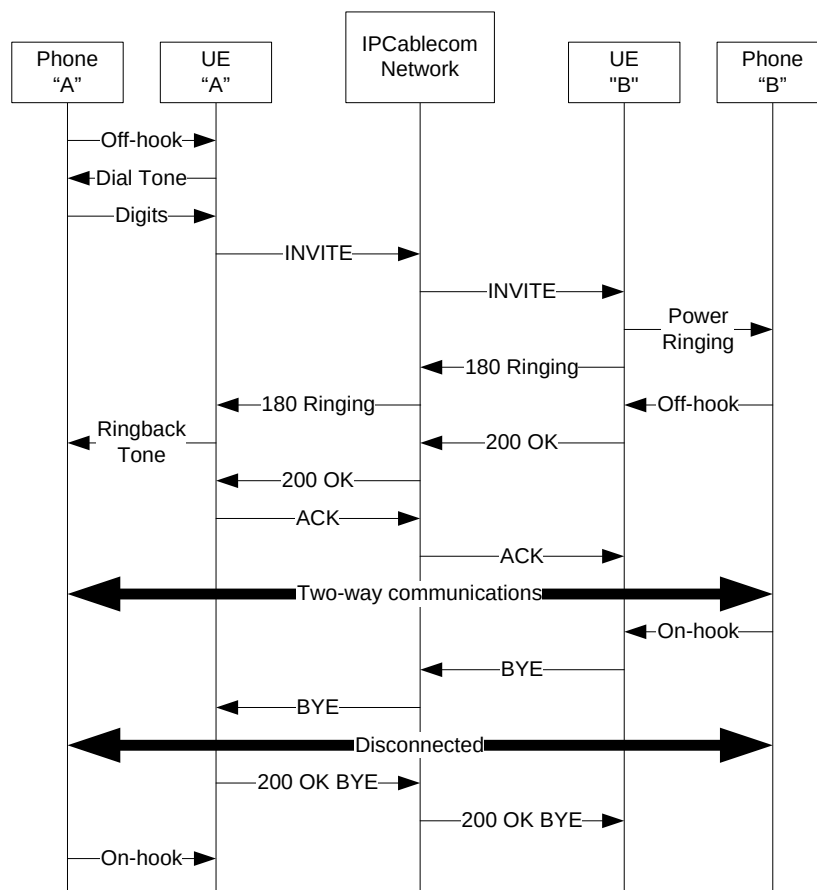


Figure 2 illustrates a call from one UE to another UE where the call is not answered. A provisionable timer (defined in Table 10) is used on the called UE to decide when to stop ringing and send back an error response to the caller.

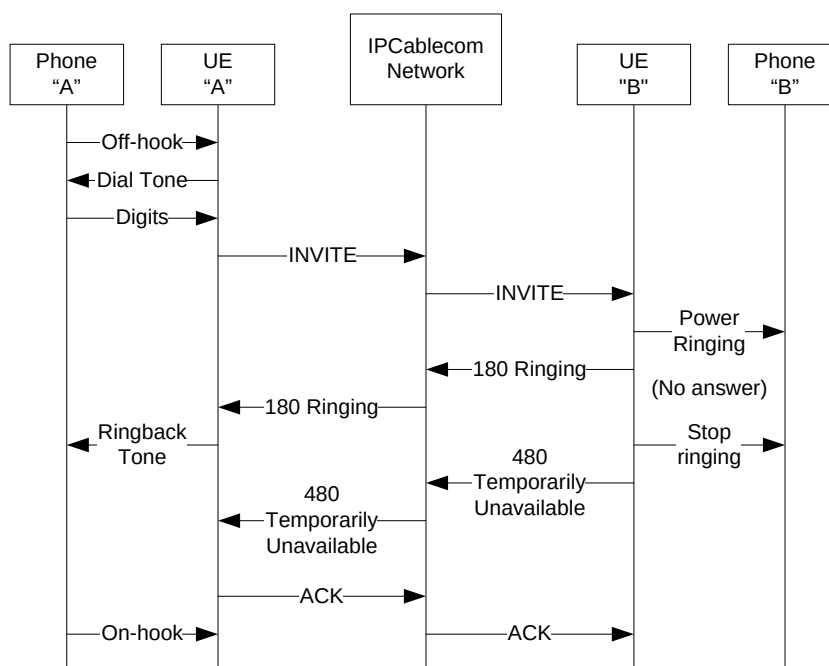


Figure 2 – Basic Call; On-Net to On-Net No Answer

Figure 3 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.

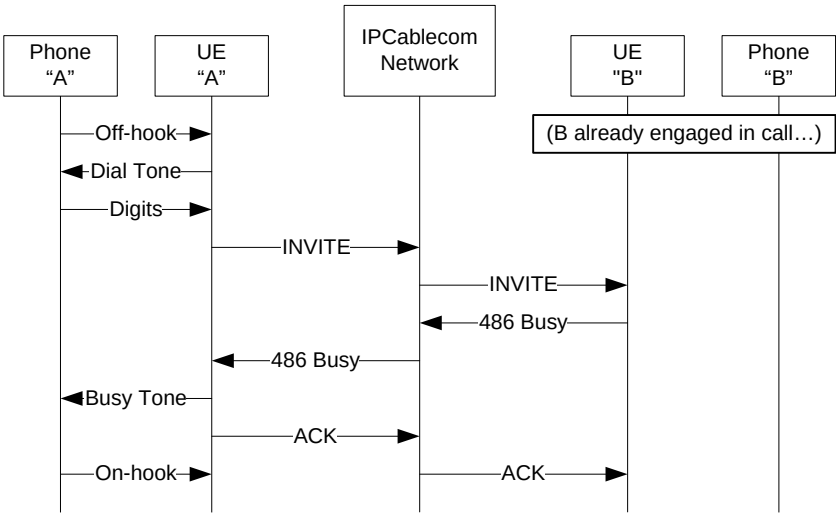


Figure 3 – Basic Call; On-Net to On-Net Busy, Call Waiting Not Applied

Figure 4 illustrates a call from a UE to a phone on the PSTN network.

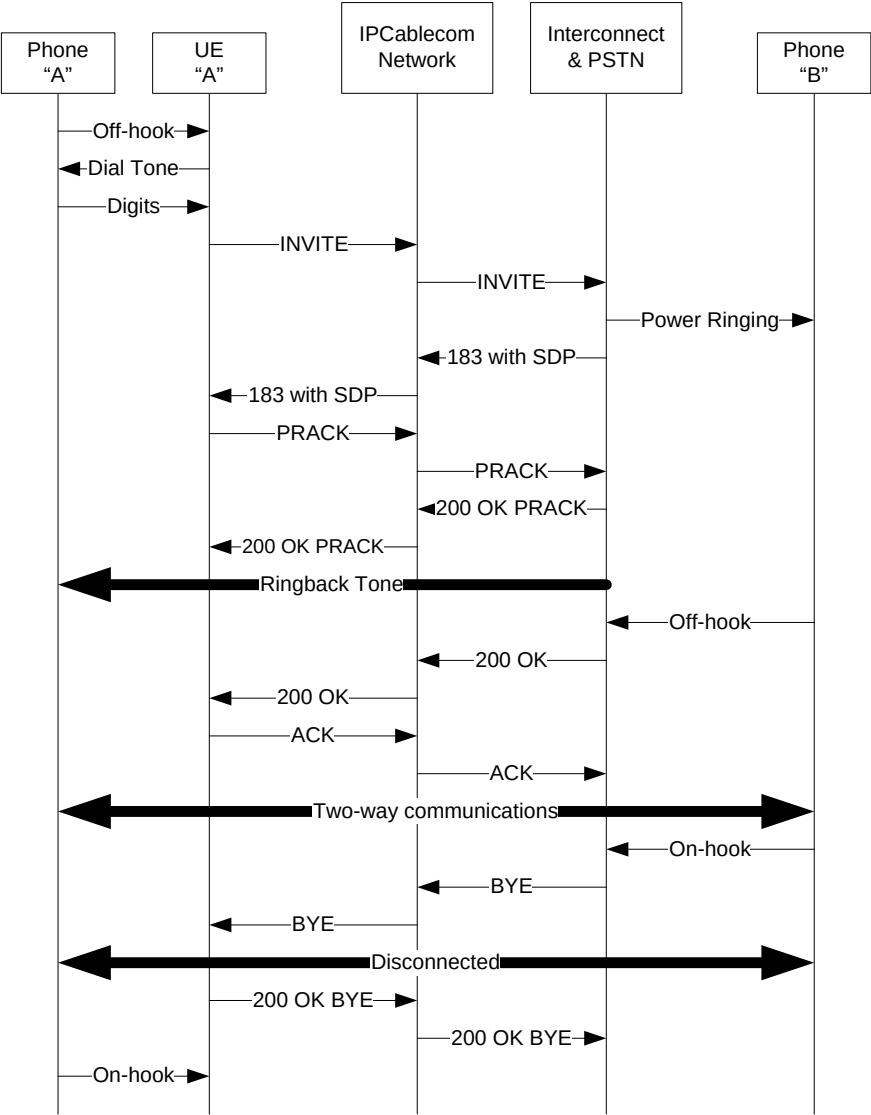


Figure 4 – Basic Call; On-Net to PSTN Off-Net

Figure 5 illustrates a call from a phone on the PSTN network to a UE (on the IPCablecom2 network).

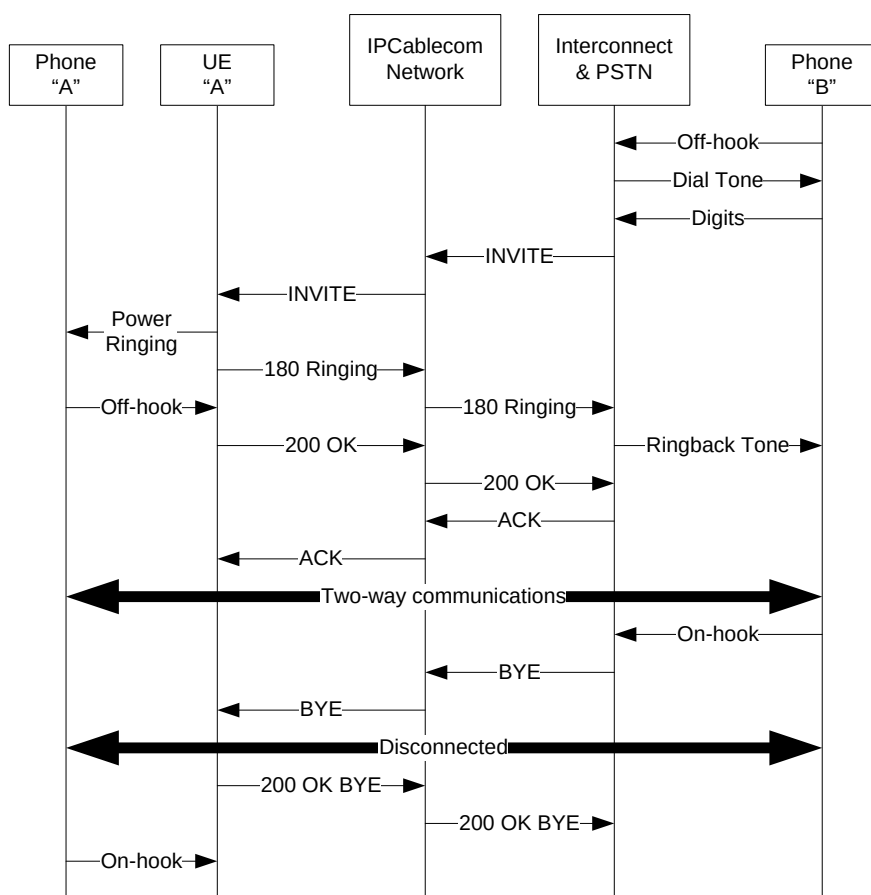


Figure 5 – Basic Call – PSTN Off-Net to On-Net

7.1.5 Held Media

There are a number of service capabilities defined in this Recommendation that require an active bidirectional media stream to be held.

For the purposes of this Recommendation, 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 Recommendation, the media attributes may be negotiated to a value other than inactive; this is discussed in clause 7.1.5.1.

This Recommendation 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 Recommendation.

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 clause.

While the requirements detailed are written with reference to the interaction between UEs, they MUST apply equally to any IPCablecom2 components between which an INVITE initiated SIP dialog can be established.

In the procedures defined within this clause, 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 that 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 clause.

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 Recommendation or [RFC 3264]. If the recipient of the hold offer SDP is not conformant to either this Recommendation or [RFC 3264] as indicated by the answering SDP, and if the controlling UE is operating using IPv4 then the controlling UE SHOULD offer an updated SDP to the other party that contains a connection data line in the media block of the SDP set to `c=IN IP4 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 Recommendation.

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 signalled. 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 the offering party.

If the UE receives an SDP offer 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 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= IN IP4 0.0.0.0 and the UE is operating using IPv4, then the UE MUST include connection data of c= IN IP4 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= IN IP4 0.0.0.0 and the UE is operating using IPv4, then the UE MUST answer with an updated SDP to the offering party and include connection data of c= IN IP4 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= IN IP4 0.0.0.0 and the UE is operating using IPv4, 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 clause, 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 in conformance with 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 bidirectional 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 clauses of this Recommendation 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 clause 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 6 illustrates the holding of an existing bidirectional unicast media stream.

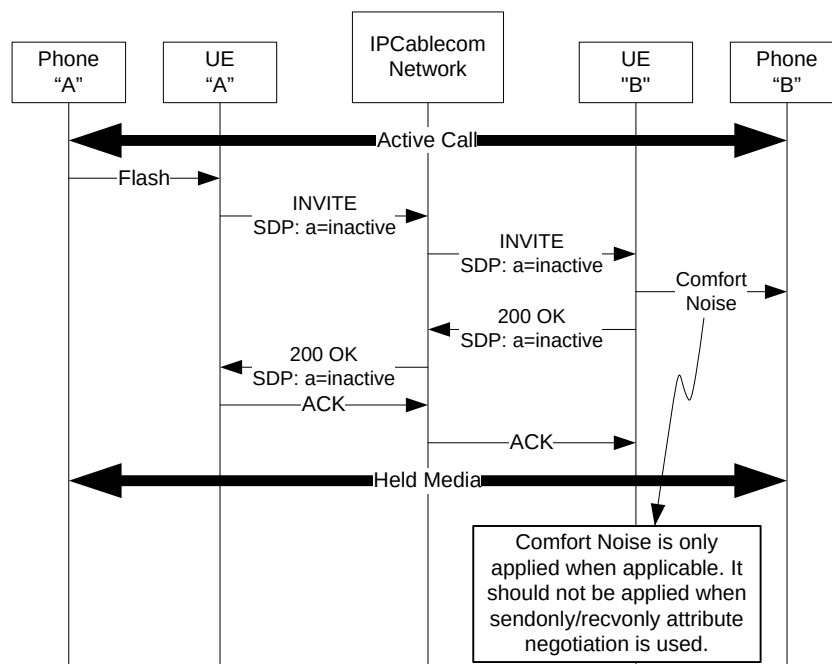


Figure 6 – Sample Held Media Message Flow

Figure 7 illustrates the use of a fallback by a UE using IPv4 to [RFC 2543] defined hold when the initial hold attempt fails because the 'held UE' does not understand the a=inactive directionality attribute.

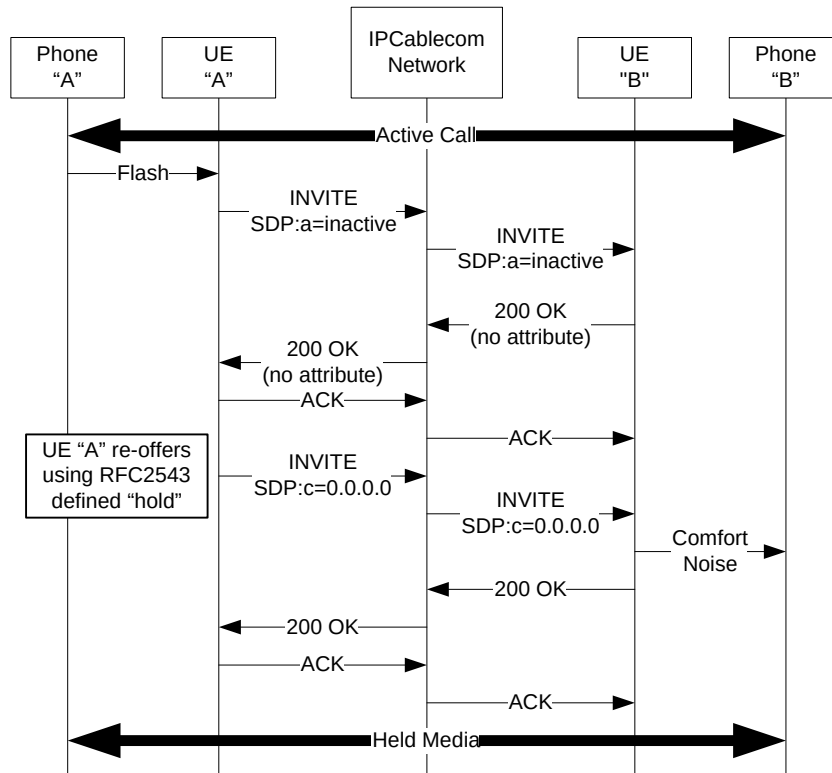


**Figure 7 – Sample Held Media Message Flow IPv4 Mode UE;
Fallback to [RFC 2543] Defined Hold**

Figure 8 illustrates the retrieval of an existing held unicast media stream.

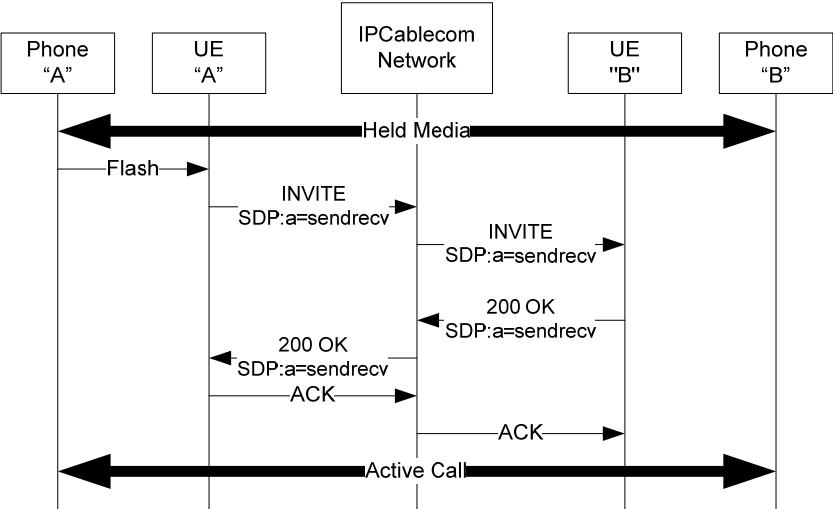


Figure 8 – Sample Held Media Message Flow; Retrieval of Held Media

Figure 9 illustrates a message flow for the transition to a dual-hold state.

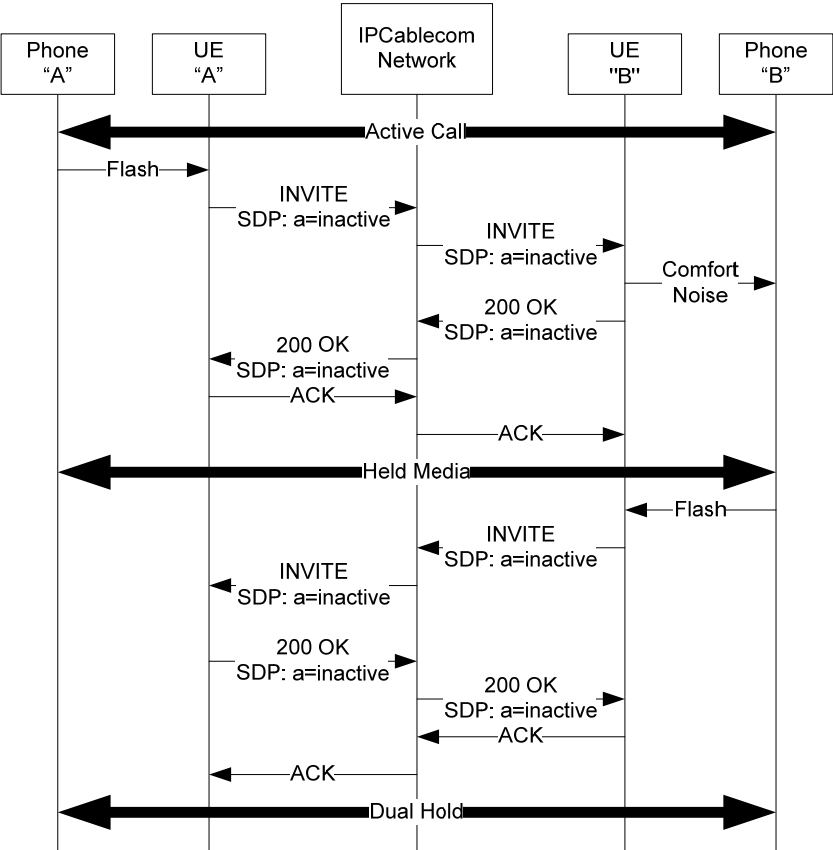


Figure 9 – Held Media; Transition to Dual-Hold

Figure 10 illustrates a message flow when one party in a dual-hold state retrieves the media.

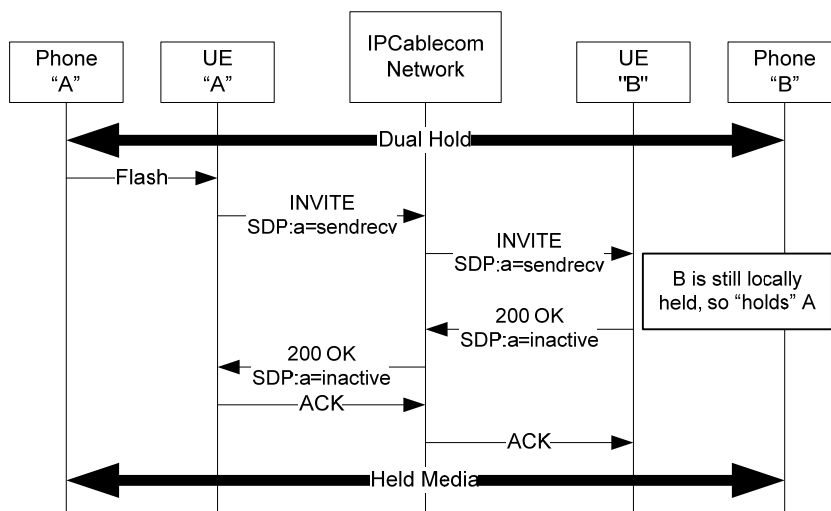


Figure 10 – Held Media; Retrieval by One Party When in Dual-Hold

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 Recommendation.

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 Recommendation.

Therefore, there are two types of announcements in IPCablecom2: tone and audio announcements.

The UE supports the analog line tones listed in referenced Telcordia, ETSI, or ITU standards for analog telephony services of the target market, physical location or operator directive within the constraints of this clause. The specific tones are provisioned and map to the tone-identifiers defined in this Recommendation.

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 the 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 Recommendation 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 clause.

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 (clause 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.

When the announcement is to be provided by a SIP enabled media server in the network which is conformant to [RFC 4240], the announcement addresses supplied in alert-info headers and contact headers **MUST** comply with the annnc-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
```

When the announcement is to be provided by a media server in the network that is accessible via other URI schemes, 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 handling of URL at the UE is defined below:

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

*** End of document ***

- 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 locally provisioned 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 Announcement Identifiers

Network initiated tone announcement addresses conveyed in the alert-info header for UE internally stored or generated tones use the URI format file:///PacketCableRST/<tone-identifier>, where the <tone-identifier> element is per Table 4.

Table 4 – Tone Announcement Identifiers

Meaning/Description	<tone-identifier>	Considerations
DTMF tones	0-9, *, #, A, B, C, D	DTMF tones (0-9, *, #, A, B, C, D)
Busy tone	bz	
Confirmation tone	cf	
Dial tone	dl	
Message waiting indicator	mwi	
Off-hook warning tone	ot	Receiver Off Hook Tone (ROH Tone) sometimes referred to as "Howler" tone.
Distinctive ringing and ringback tones (0..7)	r0, r1, r2, r3, r4, r5, r6 or r7	The provisioning process MAY define the ringing cadence for each of the r0-r7 tone identifiers. These tone identifiers may be used for distinctive caller ringback tones. They may also be used for distinctive power ring to the called party.
Ringing	rg	The value for rg is as defined for distinctive ringing pattern 1. These tone identifiers may be used for distinctive caller ringback tones. They may also be used for distinctive power ring to the called party.

Table 4 – Tone Announcement Identifiers

Meaning/Description	<tone-identifier>	Considerations
Reorder tone	ro	
Ringsplash	rs	Also known as "Reminder ring" is a burst of ringing that may be applied to the physical forwarding line (when idle) to indicate that a call has been forwarded and to remind the user that a Call Forwarding subfeature is active.
Ring back tone	rt	
Stutter dial tone	sl	Also called Recall Dial Tone
Call waiting tones	wt1, wt2, wt3, wt4	The provisioning process MAY define the tone pattern used for each of the wt1-wt4 tone identifiers.
DTMF tones wildcard	X	The DTMF tones wildcard matches any DTMF digit between 0 and 9

Network generated tones or Network directed tone generation could use the same <tone-identifier> convention in the alert-info header for signaling of the desired tone.

7.1.6.4 Audio Announcement Identifiers

IPcablecom2 supports the following basic categories of audio 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 – Audio 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 cannot accept your call at this time. Please hang up and try again later."	Annc-Denied
No Circuit	"Your call cannot be completed at this time. Please hang up and try again later."	Annc-NoCircuit

Table 5 – Audio Announcement Identifiers

Category/Condition	Announcement	Announcement Identifier
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 announcement 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
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-Denied
603 Decline	Annc-Denied

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 11 illustrates this process.

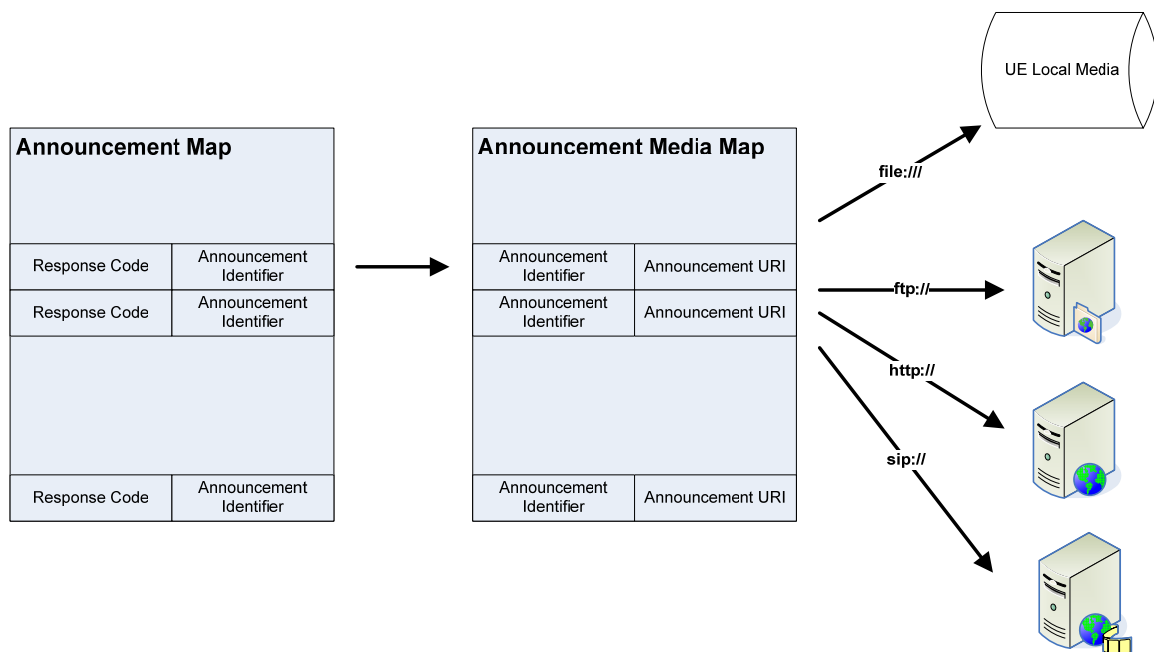


Figure 11 – Announcement Media Mapping

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

7.1.6.5 Management of Announcement Media Cache

As described in clause 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 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.

7.1.6.6 Feature Execution

This clause 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 locally provisioned tone, a local announcement or a network announcement, depending on its provisioning:

Locally Provisioned Tone. In the absence of any network announcement provisioning, the UE MUST play the locally provisioned 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 cannot map the prompt-url constructed by the UE to a valid announcement media), the UE MUST apply the locally provisioned 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

This is based on restrictions on header in some SIP requests and responses. The Contact header can be used in places where alert-info cannot be used, and is more appropriate when the AS is making use of an alternate server for a media session rather than when the AS is directing the UE to play out locally.

- The Application Server may provide an announcement address in a Contact header returned with a non-2xx final response message.
- The application server provides the UE with the announcement address as 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 clause.

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 indicating 487 Request Terminated.

7.1.6.7 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	Config. Server Requirement
Announcement Resource	Xsd:anyURI for media server	Non-volatile on AS, Volatile on UE	Per network	Config. Server	Config. Server	AS, UE	Mandatory

Table 7 – Announcement Feature Data

Data	Type	Persistence	Scope	Stored by	Written by	Read by	Config. Server Requirement
Announcement Domain	Character string (Domain name)	Non-volatile on, Volatile on UE	Per network	Config. Server	Config. Server	AS, UE	Mandatory
Announcement Path	Character string (Network file path)	Non-volatile on, Volatile on UE	Per network	Config. Server	Config. Server	AS, UE	Mandatory
Announcement mime type	RFC 2045	Non-volatile on, Volatile on UE	Per network	Config. Server	Config. Server	AS, UE	Mandatory
Announcement map	Variable length array of Announcement map entries	Volatile on UE	Per network	Config. Server	Config. Server	UE	Mandatory
Announcement map entry.Response code	Integer	Volatile on UE	Per network	Config. Server	Config. Server	UE	Mandatory
Announcement map entry.Announcement URI	Character string identifying uri for response code	Volatile on UE	Per network	Config. Server	Config. Server	UE	Mandatory
Announcement media map	Variable length array of Announcement Media Map entries	Volatile on UE	Per network	Config. Server	Config. Server	UE	Mandatory
Announcement media map entry.Announcement Identifier	Character string identifying the Announcement identifier	Volatile on UE	Per network	Config. Server	Config. Server	UE	Mandatory
Announcement media map entry.Announcement URI	Character string identifying URI for Announcement Identifier	Volatile on UE	Per network	Config. Server	Config. Server	UE	Mandatory
Announcement media cache maximum-age	Integer (Seconds)	Volatile on UE	Per Network	Config. Server	Config. Server	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	Config. Server Requirement
Local Media Table	Character string Variable length array of Media entries	Volatile on UE	Per network	Config. Server	Config. Server	UE	Mandatory
Media.URI	Character string File:/// URI identifying locally stored media	Volatile on UE	Per network	Config. Server	Config. Server	UE	Mandatory
Media.Type	Media Type (from set of bell cadence, tone generator specification, or audio file)	Volatile on UE	Per UE	UE	UE	UE	Not Applicable
Media.Data	The actual media data, format depends on Media.Type	Volatile on UE	Per UE	UE	UE	UE	Not applicable

Local media types supported are audio files, ringing signals or tones. Audio files identified by the file type (for example, mp3, wav) are implementation dependent. Ringing signals and tones are defined by Telcordia, ETSI or ITU standards or Recommendations.

7.1.6.8 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.

7.1.6.9 Support for Language Preference (Optional)

IPCablecom2 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 subclauses.

7.1.6.9.1 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 clause 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.

7.1.6.9.2 Behaviour 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 locally provisioned 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 clause 7.1.6.6. 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 clause 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 (locally provisioned) 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 clause 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 locally provisioned 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.

7.1.6.9.3 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	Config. Server Requirement
Language	Character String	volatile in UE	Per public user ID	Config. Server	Config. Server	UE	Mandatory

7.1.6.9.4 Behaviour of Application Server for Language Preference

All language preference supported by network elements such as an Application Server 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.

7.1.6.9.5 Behaviour 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 behaviour 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.

7.1.6.10 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 12 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.

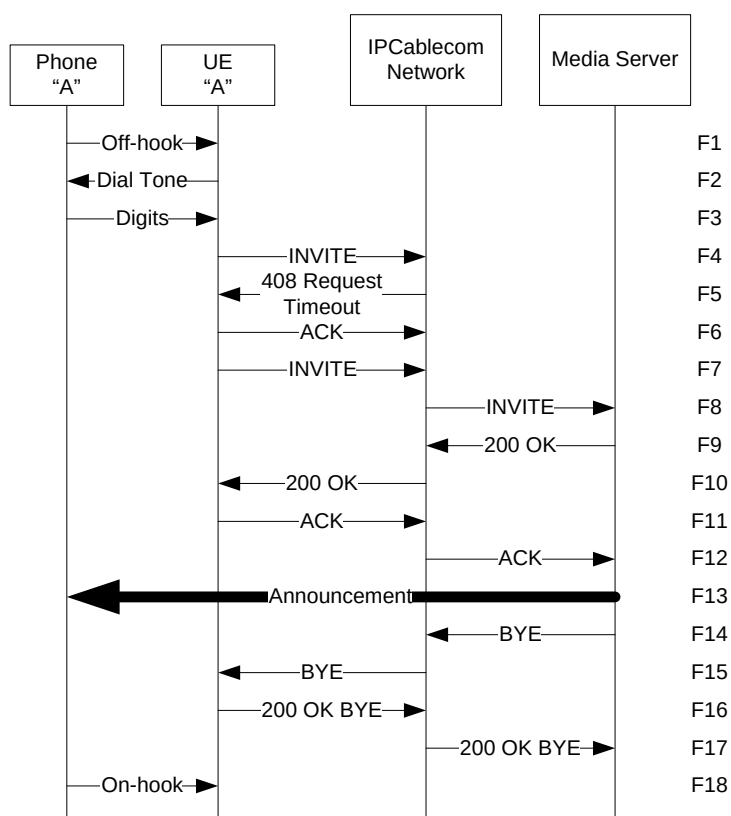


Figure 12 – Media Serving; UE INVITE for Network Announcement

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 13 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, the Application Server provides the reference to the appropriate announcement in a Contact header.

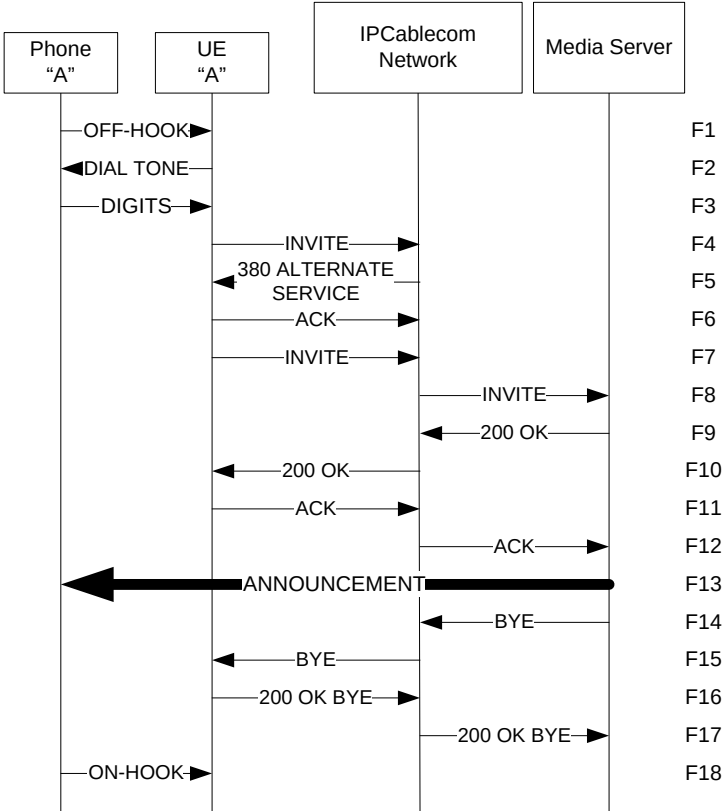


Figure 13 – Media Serving; Server Redirecting UE

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

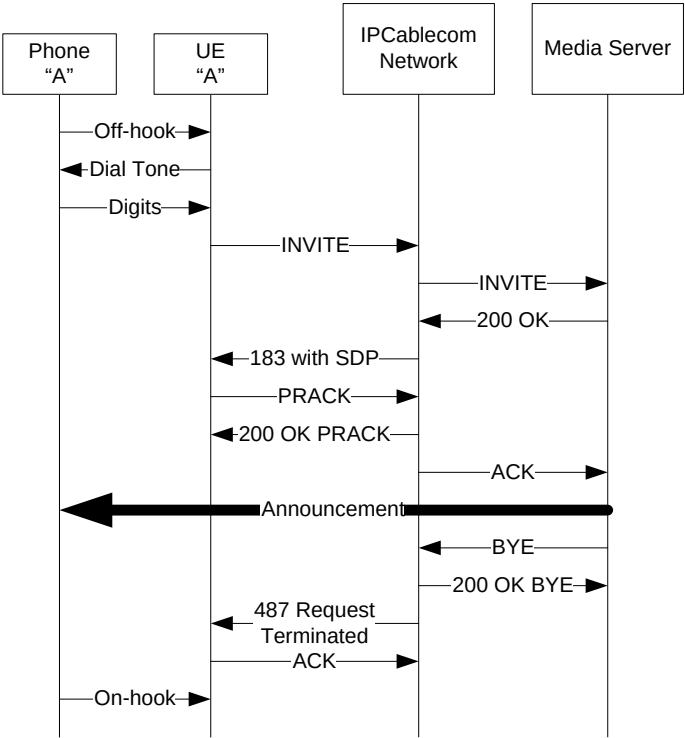


Figure 14 – Media Serving; Server Providing Announcement with Early Media