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B. Foster  
F. Andreassen  
Cisco Systems  
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## Basic Media Gateway Control Protocol (MGCP) Packages

### Status of this Memo

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### IESG Note

This document is being published for the information of the community. It describes a non-IETF protocol that is currently being deployed in a number of products. Implementers should be aware of RFC 3525 [37], which was developed in the IETF Megaco Working Group and the ITU-T SG16, and is considered by the IETF and ITU-T to be the standards-based (including reviewed security considerations) way to meet the needs that MGCP was designed to address. The IETF Megaco Working Group and the ITU-T Study Group 16 are developing extensions to RFC 3525 [37] that for functions of the type in addressed in this document.

### Abstract

This document provides a basic set of Media Gateway Control Protocol (MGCP) packages. The generic, line, trunk, handset, RTP, DTMF (Dual Tone Multifrequency), announcement server and script packages are updates of packages from RFC 2705 with additional explanation and in some cases new versions of these packages. In addition to these, five new packages are defined here. These are the signal list, resource reservation, media format, supplementary services and digit map extension packages.

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## 1. Introduction

This document provides a basic set of Media Gateway Control Protocol (MGCP) packages. The generic, line, trunk, handset, RTP, DTMF, announcement server and script packages are updates of packages from RFC 2705 [38] with additional explanation and in some cases new versions of these packages. In addition to these, five new packages are defined here. These are the signal list, resource reservation, media format, supplementary services and digit map extension packages.

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14, RFC 2119 [31].

### 1.1. List of Packages

The basic set of packages specified in this document is for use with MGCP 1.0 as defined in [1]. Included are the following packages:

Package	Name
Generic Media Package	G
DTMF package	D
Trunk Package	T
Line Package	L
Handset Package	H
Supplementary Services Package	SST
Digit Map Extension	DM1
Signal List Package	SL
Media Format Package	FM
RTP Package	R
Resource Reservation Package	RES
Announcement Server Package	A
Script Package	Script

### 1.2. Changes to Existing RFC 2705 Packages

#### 1.2.1. Change in signal types

MGCP 1.0, as defined in RFC 2705 [38] (and now updated in [1]), provided some additional clarification on the meaning of On-Off (OO) signals compared to earlier versions of MGCP. This led to some inconsistency in some of the signal definitions in the accompanying packages in RFC 2705 [38]. This has been corrected in the packages that are included here by changing some of the signals from type On-Off to type Time-Out (TO).

#### 1.2.2. Operation Complete and Operation Failure

Another change made to improve consistency and interoperability was to add the "operation complete" and "operation failure" events in packages where there are TO signals defined, but where the "operation complete" and "operation failure" events were not previously included as part of the package. By definition, all packages that contain

Time-Out type signals now contain the "operation failure" ("of") and "operation complete" ("oc") events as defined in [1], irrespective of whether they are provided as part of the package description or not.

If a package without Time-Out signals contains definitions for the "oc" and "of" events, the event definitions provided in the package may over-ride those indicated here. Such practice is however discouraged and is purely allowed to avoid potential backwards compatibility problems.

It is considered good practice to explicitly mention that the "oc" and "of" events are supported in accordance with their default definitions. If no definition is included in the package, the default syntax and semantics are assumed.

Please refer to [1] for additional details on these events.

### 1.2.3. Package Versions

The generic, line, trunk, handset, RTP, DTMF, announcement server and script packages included in this document are new versions of packages that were previously contained in RFC 2705 [38]. The updated base MGCP 1.0 specification [1] provides an optional capability of auditing package versions. Any gateway that implements versioned packages SHOULD also implement this option.

### 1.2.4. Event Definitions, Aliases and Interoperability Issues

Some event definitions or clarifications of previous event definitions have also been added in order to improve interoperability.

In some cases, events have aliases either in the same or in other packages and a recommendation has been made for the use of alternates by Call Agents for future implementations. For maximum interoperability, gateways MUST still implement these events (in fact they MUST always implement all of the events, signals, etc. in a package).

Some events that were previously defined require specific provisioning in both the gateway and the Call Agent in order to allow for interoperability. In those cases, a warning to that affect has been included.

### 1.2.5. New Events

In some cases, new events have been added to existing packages. Any changes to existing packages of course have resulted in the package version number being updated from unversioned (version 0) to version 1.

### 1.3. New Packages and Excluded Packages

Two packages from RFC 2705 [38] have not been included. These are the "MF" and the "NAS" packages. These packages are still valid as are all unversioned (version 0) packages defined in RFC 2705 [38]. The reason these packages were not included are:

- \* The original MF package had no defined way to outpulse MF digits so that MF CAS is now provided by other packages (i.e., the "MS", "MO" and "MD" packages) in a separate document.
- \* The "N" package, as defined in RFC 2705 [38], was incomplete. A new MGCP "NAS" package has been developed and provided in a separate document.

New packages have also been included beyond what was included in RFC 2705 [38]. These are the signal list, resource reservation, media format, supplementary services and digit map extension packages. The Resource Reservation ("RES") and Media Format ("FM") packages in particular are different from other packages in this document in that they contain new LocalConnectionOptions. This is allowed by the new extension rules in [1]. Future packages of this type MUST use a package prefix in front of local connection options ("`<package-name>/<Local Connection Option>`") so as to avoid name-space problems. However because of the timing of the arrival of these packages relative to updating MGCP 1.0, this was not done for the "RES" and "FM" packages. The resulting new local connection options have been registered with IANA. For future cases where a package prefix is included, only the package name needs to be registered.

## 2. Packages

For those packages that involve MGCP events, the terms "signal" and "event" are used to differentiate a request from a Call Agent to a Media Gateway to apply an event ("signal"), from the request for the detection of an "event" that occurs on the Media Gateway and is "Notified" to the Call Agent.

For packages that involve events and signals, the tables contain five columns:

Symbol: the (package) unique symbol used to identify the event.

Definition: a short description of the event.

R: an x appears in this column if the event can be requested by the Call Agent. Alternatively, one or more of the following symbols may appear. An "S" is included if the event-state may be audited. A "C" indicates that the event can be detected on a connection, and a "P" indicates the event is persistent.

S: if nothing appears in this column for an event, then the event cannot be signaled by the Call Agent. Otherwise, the following symbols identify the type of event:

- \* OO On/Off signal
- \* TO Time-Out signal.
- \* BR Brief signal.

In addition, a "C" will be included if the signal can be generated on a connection.

Duration: specifies the default duration of TO signals. If a duration is left unspecified, then the default timeout will be assumed to be infinite, unless explicitly noted in the description of the signal. A duration may also be declared as being variable in a case where signals involve complex sequencing (e.g., scripts or digit out-pulsing) where the amount of time may vary with either processing time or the signaling environment.

Default time-out values may be over-ridden by the Call Agent for any Time-Out event defined in this document (with the exception of those that have a default value of "variable") by a "to" signal parameter which specifies the timeout value in milliseconds (see [1]). The following example indicates a timeout value of 20 seconds:

```
S: sst/cw(to=20000)
```

As indicated in [1]: by default, a supplied time-out value MAY be rounded to the nearest non-zero value divisible by 1000, i.e., whole second. However, individual signal definitions within a package may define other rounding rules.

Note that Time-Out signals that involve other parameters still allow the use of the "to" signal parameter e.g.:

```
S: T/sit(1,to=3000)
```

The order of the "to" parameter relative to the other parameters is not important.

Note: as per [1], On-Off (OO) signals are parameterized with "+" (meaning turn on) or "-" (meaning turn off). If the parameter is missing, the default is to turn on the signal. Unlike Time-Out signals, On-Off signals do not stop when an event occurs.

Other than the "to" parameter for Time-out (TO) signals and the "+" and "-" for On-Off (OO) signals, signals and events in the packages in this document do not have parameters unless explicitly indicated in the description of the event for that package.

In some of the signal definitions below, specific tone definitions are provided even though actual frequencies may vary from country to country.

## 2.1. Generic Media Package

Package Name: G

Version: 1

The generic media package groups the events and signals that can be observed on several types of endpoints, such as trunk gateway endpoints, access gateway endpoints or residential gateway endpoints.

Symbol	Definition	R	S	Duration
cf	Confirm Tone		BR	
cg	Congestion Tone		TO	infinite
ft	Fax Tone	x		
it	Intercept Tone		TO	infinite
ld	Long Duration Connection	C		
mt	Modem Tone	x		
oc	Operation Complete	x		
of	Operation Failure	x		
pat(###)	Pattern Detected	x	OO	
pt	Preemption Tone		TO	infinite
rbk(...)	Ringback		TO,C	180 seconds
rt	Ringback Tone		TO,C	180 seconds

New events added to this package from the previously unversioned package: "oc"

Changes: "it" and "pt" signals changed from OO to TO.

Note that default time-out values may be over-ridden by the Call Agent for any Time-Out signal defined in this package by a "to" signal parameter. Refer to section 2 of this document, as well as [1] for details.

The events and signals are defined as follows:

Confirmation Tone (cf):

This is also referred to as "positive indication tone" in ITU-T E.182. In North America, Confirmation Tone uses the same frequencies and levels as dial tone (350 and 440 Hertz) but with a cadence of 0.1 second on, 0.1 second off, repeated three times. See GR-506-CORE [7] Section 17.2.4. It is considered an error to try and play confirmation tone on a phone that is on-hook and an error MUST consequently be returned when such attempts are made (error code 402 - phone on-hook).

Congestion Tone (cg):

Refer to ITU-T E.180 [8] and E.182 [10]. This maps to re-order tone in North America (refer to GR-506-CORE [7] Section 17.2.7).

Fax Tone (ft):

The fax tone event is generated whenever a fax call is detected by the presence of V.21 fax preamble. The fax tone event SHOULD also be generated when the T.30 CNG tone is detected. See ITU-T Recommendations T.30 [21] and V.21 [22].

Intercept Tone(it):

This is a country specific tone as defined in ITU-T E.180 Supplement 2 [9].

Long Duration Connection (ld):

The "long duration connection" is detected when a connection has been established for more than a provisioned amount of time. The default value is 1 hour.

This event is detected on a connection. When no connection is specified as part of the request, the event applies to all connections for the endpoint, regardless of when the connections are created. The "all connections" wildcard (see [1]) may also be used for this case, and is in fact preferred for consistency. In

either case, the name of the connection on which the event was detected will be included when the event is observed, e.g.:

G/ld@0A3F58

Modem Tone (mt):

Indicates V.25 Answer tone (ANS) with or without phase reversals or V.8 Modified Answer Tone (ANSam) tone with or without phase reversals. Note that this implies the presence of a data call. Also note that despite the name of the event, devices other than modems may generate such tones, e.g., a fax machine.

Operation Complete (oc):

The standard definition of operation complete [1].

Operation Failure (of):

The standard definition of operation failure [1].

Pattern Detected (pat(###)):

This event requires special provisioning that needs to be agreed on between the Call Agent and media gateway in order to ensure interoperability. It is retained in order to maintain backwards compatibility with version 0 of the "G" package. This event MUST be parameterized with a decimal numeric value from 0 to 999 specifying the pattern to detect. When reported, the pattern is also included as a parameter.

Preemption Tone (pt):

This is a country specific tone and is defined in ITU-T E.180 Supplement 2 [9].

Ringback (rbk(connectionID)):

This is an alias for "rt@connectionID" and is included here for backwards compatibility only. It is recommended that Call Agents use "rt@connectionID" instead of "rbk(connectionID)" for ring-back over a connection for new implementations. Although the ringback signal is applied on a connection, the "rbk" signal does not support the "@connection" syntax. When the signal is requested, it MUST be parameterized with a connection-ID or a connection-ID wildcard as specified in [1].

Ringback Tone (rt):

Refer to ITU-T E.180 [8] and ITU-T E.182 [10]. Also referred to as ringing tone - a tone advising the caller that a connection has been made and that a calling signal is being applied to the called party or service point. In North America, this tone is a combination of two AC tones with frequencies of 440 and 480 Hertz and levels of -19 dBm each, to give a combined level of -16 dBm.

The cadence for Audible Ring Tone is 2 seconds on, followed by 4 seconds off. See GR-506-CORE [7] - LSSGR: SIGNALING, Section 17.2.5.

This signal can be applied directly to an endpoint or alternatively on a connection using the syntax "rt@connectionID". When the ringback signal is applied to an endpoint, it is considered an error to try and play ringback tone if the endpoint is considered on-hook, and an error MUST consequently be returned when such attempts are made (error code 402 - phone on-hook). When the ringback signal is applied to a connection, no such check is to be made.

Note that as specified in [1], signals requested on a connection MUST be played regardless of the connection mode. For example, in a call-waiting situation, ringback tone may be played on a connection in "inactive" mode.

## 2.2. DTMF package

Package name: D

Version: 1

Symbol	Definition	R	S	Duration
0	DTMF 0	x	BR	
1	DTMF 1	x	BR	
2	DTMF 2	x	BR	
3	DTMF 3	x	BR	
4	DTMF 4	x	BR	
5	DTMF 5	x	BR	
6	DTMF 6	x	BR	
7	DTMF 7	x	BR	
8	DTMF 8	x	BR	
9	DTMF 9	x	BR	
#	DTMF #	x	BR	
*	DTMF *	x	BR	
A	DTMF A	x	BR	
B	DTMF B	x	BR	
C	DTMF C	x	BR	
D	DTMF D	x	BR	
DD(..)	DTMF Tone Duration	x	TO	3 seconds
DO(..)	DTMF OO Signal		OO	
L	Long Duration Indicator	x		
oc	Operation Complete	x		
of	Operation Failure	x		
T	Interdigit Timer	x	TO	16 seconds
X	DTMF Tones Wildcard, match any digit 0-9	x		

Changes from the previous version of the package: events "dd", "do", "oc" were added.

Note that DTMF tones including the DTMF tones wildcard can use the eventRange notation defined in [1] when requesting events, e.g., "D/[0-9](N)".

Note that default time-out values may be over-ridden by the Call Agent for any Time-Out signal defined in this package by a "to" signal parameter. Refer to section 2 of this document, as well as [1] for details.

The events are defined as follows:

DTMF tones (0-9,#,\*,A,B,C,D):

Detection and generation of DTMF tones is described in GR-506-CORE [7] - LSSGR: SIGNALING, Section 15. Note that it is considered an error to try and play DTMF tones on a phone that is on-hook and an error MUST consequently be returned when such attempts are made (error code 402 - phone on-hook). The event codes can be specified in a digit map. When requested as a signal, as per GR-506-CORE [7], section 15, a minimum tone duration of 50 ms will be followed by a minimum interdigit silence period of 45 ms, i.e., if requested in a signal list such as "S: sl/s(d/5,d/6,d/7)", then interdigit timing requirements will be satisfied.

Note that some types of endpoints, such as announcement endpoints, MAY allow detection and/or generation of a DTMF tone over a connection. However, this requires consistent provisioning between the Call Agent and announcement server (it is not required in order to be compliant with the DTMF package).

DTMF Tone Duration (dd(dg=<tone>,to=<time>,su=<TrueOrFalse>)):

This event can be used to indicate if/when the specified <tone> has a duration greater than the <time> value indicated (and is reported once the duration is exceeded). The parameters can be supplied in any order. The value of <tone> can be any of the DTMF tone symbols (without including the package name) specified in the DTMF package (including X in the case of events, but not signals). If this parameter is absent, any DTMF tone that occurs will be reported. The parameter <time> is in milli-seconds and may be rounded to the nearest 10 ms by the gateway. The minimum value of <time> that can be requested when requesting an event is 40 ms. When requesting a signal, the minimum value of <time> that can be requested is 50 ms. The maximum value of <time> that can be requested for either an event or a signal is 60000 ms. If the "to=<time>" parameter is absent when requested as an event, the event will report the full duration (up to 60000 ms) of the tone when the tone is completed. When reported as an ObservedEvent, both parameters are always supplied. In this case, <tone> is the actual tone detected and <time> is either:

- \* The <time> specified in the request (possibly rounded), or
- \* If the request did not contain a "to=<time>" parameter, the full duration of the tone.

The parameter "su" MAY be included when this is requested as an event (but is not reported). This parameter is used to indicate whether or not the DTMF digits requested should be suppressed

in-band when it is requested. Possible values are "true", indicating that in-band DTMF should be suppressed and "false" indicating that DTMF should continue to be passed in-band. The default value of the parameter, if missing, is "false". The "su" parameter MUST NOT be included when requesting "D/dd" as a signal.

When used as a signal, "dd" provides the ability to generate a DTMF tone as a TO signal. When applied as a signal, an additional 50 ms of silence will be tacked onto the end before the operation complete occurs, i.e., "S: dd(dg=5,to=2500)" will play the DTMF tone for the number "5" for 2.5 seconds, followed by 50 ms of silence period. The operation complete (if requested) will be notified after the silence interval occurs. Any value from 50 ms to 60000 ms can be requested. Gateways generating or detecting the tone may round off the requested time to the nearest 10 ms.

The "dd" event can be used in place of the "long duration" event in order to detect a digit pressed for longer than 2 seconds. For example, in order to detect if a user presses the long "#" for longer than 2 seconds, a request could be made with the RequestedEvents line "R: d/dd(N)(dg=#,to=2000)". The resulting ObservedEvents line would be "O: d/dd(dg=#,to=2000)".

Suppose instead, that the RequestedEvents line contains

```
R: d/[0-9*#],d/dd
```

Suppose the user then pushes the "#" for 2.5 seconds. In this case, two events will be notified:

```
O: d/#
```

when the "#" key is first pressed, and

```
O: d/dd(dg=#,to=2500)
```

when the "#" key is finally released.

DTMF OO Signal (do(dg=<tone>,<on-or-off>)):

This signal is used to generate a DTMF tone as an on-off signal. The <tone> parameter is any of the symbols for a specific tone in the DTMF package (i.e., "0" to "9", "A", "B", "C", "D", "\*", or "#"). The <on-or-off> indicator is "+" for on and "-" for off as per [1]. The <tone> parameter MUST be supplied, otherwise a return code of 538 - "Event/signal parameter error" will be provided in the response. If the <on-or-off> parameter is missing, the default is to turn the signal on as usual (i.e., "+" is the default). The order of the parameters is not significant

since "+" and "-" are reserved characters and are easily distinguished from the <tone> parameter.

Long Duration Indicator (l):

The "long duration indicator" is observed when a DTMF signal is produced for a duration larger than two seconds. In this case, the gateway will detect two successive events: first, when the signal has been recognized, the DTMF signal, and then, 2 seconds later, the long duration signal.

Operation Complete (oc):

This is the standard definition of operation complete [1].

Operation Failure (of):

This is the standard definition of operation failure [1].

Timer (t):

Timer T can be used as an event or as a time-out (TO) signal. As a signal, its only behavior is the normal characteristics of a "TO" signal as defined in [1] (i.e., if no event occurs before the time-out, an operation complete event will be generated).

As an event, Timer T is a digit input timer that can be used in two ways:

- \* When timer T is used with the accumulate according to digit map action, the timer is not started until the first DTMF tone is entered, and the timer is restarted after each new DTMF tone is entered until either a digit map match or mismatch occurs. In this case, timer T functions as an inter-digit timer as illustrated by:

R: D/[0-9T](D)

- \* When timer T is used without the "accumulate according to digit map" action, the timer is started immediately and simply cancelled (but not restarted) as soon as a DTMF tone is entered. In this case, timer T can be used as an inter-digit timer when overlap sending is used, as in:

R: D/[0-9](N), D/T(N)

When used with the "accumulate according to digit map" action, timer T takes on one of two values, T-partial or T-critical. When at least one more symbol is required for the "current dial string" to match any one of the patterns in the digit map, timer T takes on the value T-partial, corresponding to partial dial timing. If a timer is all that is required to produce a match, timer T takes

on the value T-critical corresponding to critical dial timing. When timer T is used without the "accumulate according to digit map" action, timer T takes on the value T-critical. The default value for T-partial is 16 seconds and the default value for T-critical is 4 seconds. The provisioning process may alter both of these. If timer T is not used, then inter-digit timing will not be performed.

The following examples illustrate this. Consider the digit map:

```
(xxxxxxx|x11T)
```

and assume that DTMF and the timer T is accumulated according to digit map. At the first DTMF input, say "4", timer T is started with a value of T-partial since at least one more symbol is required. If "1" is then input, it leads to a restart of timer T with a value of T-partial again. If "1" is now input again, we have a current dial string of "411" and a timer is now all that is required to produce a match. Hence timer T is now restarted with value T-critical.

Finally, consider the following subtle examples (all assuming DTMF and timer T being accumulated according to digit map):

The digit map

```
(1[2-3T].)
```

will match immediately on the input "1" since zero or more matches of the range are specified.

The digit map

```
(1[2-3].T)
```

and an input of "1" will lead to timer T being set to T-critical.

A digit map of

```
(1[2-3]T.)
```

and an input of "1" will lead to timer T being set to T-partial. Furthermore, upon subsequent input of "2" or "3" a perfect match will be triggered immediately since timer T is completely irrelevant.

## DTMF Tones Wildcard (X):

The DTMF tones wildcard matches any DTMF digit between 0 and 9. The actual event code generated will however be the event code for the digit detected. The DTMF tones wildcard is often used to detect DTMF input to be matched against a digit map.

## 2.3. Trunk Package

Package Name: T  
Version: 1

Symbol	Definition	R	S	Duration
as	Answer Supervision	x	BR	
bl	Blocking		BR	
bz	Busy		TO	30 sec.
col	Continuity Tone (go tone, or return tone)	x	TO	3 sec.
co2	Continuity Test (go tone, or return tone in dual tone procedures)	x	TO	3 sec.
ct(...)	Continuity Transponder		OO	
lb	Loopback		OO	
nm	New Milliwatt Tone	x	TO	3 sec
mm	Newest Milliwatt Tone	x	TO	3 sec
oc	Operation Complete	x		
of	Operation Failure	x		
om	Old Milliwatt Tone	x	TO	3 sec
pst	Permanent Signal Tone		TO	infinite
qt	Quiet Termination		TO	infinite
ro	Reorder Tone	x	TO	30 sec.
sit(#)	Special Information Tone	x	TO	2 sec. (see notes)
tl	Test Line	x	TO	infinite
tp(###)	Test Pattern	x	TO	3 sec
zz	No Circuit	x	TO	2 sec

New events added to this package from the previously unversioned package: "bz", "ct", "mm", "oc", "pst", "qt", "sit", and "tp".

Changes in event types: "col", "co2", "nm", "om", "tl", "zz" signals changed from OO to TO; "as" and "bl" changed from OO to BR.

Note that default time-out values may be over-ridden by the Call Agent for any Time-Out signal defined in this package by a "to" signal parameter. Refer to section 2 of this document, as well as [1] for details.

The definition of the trunk package events are as follows:

Answer Supervision (as):

This event is used to indicate the occurrence answer supervision. In most cases, it is a result of a steady off-hook in response to a call request. This event is included for backwards compatibility with the previous version of the package. The preferred alternative is to use the "answer" event in the appropriate CAS packages [34] (Note: check the details on the use of "answer" in the particular CAS package; in most cases "answer" as an event is an indication of a steady off-hook regardless of whether or not it is an indication of answer supervision). For details on when answer supervision is appropriate refer to [5].

Blocking (bl):

This event is used to indicate an incoming off-hook for the purposes of blocking a one-way trunk in CAS trunks. This event is included for backwards compatibility with the previous version of the package. The preferred alternative is the "block" event in the appropriate CAS packages [34].

Busy Tone (bz):

Refer to ITU-T E.180 [8]. In North America, station Busy is a combination of two AC tones with frequencies of 480 and 620 Hertz and levels of -24 dBm each, to give a combined level of -21 dBm. The cadence for Station Busy Tone is 0.5 seconds on, followed by 0.5 seconds off, then repeating. See GR-506-CORE [7]- LSSGR: SIGNALING, Section 17.2.6.

Continuity Tone (co1):

A tone at 2010 Hz (see section 3.1.1.3 of [2]). When generated as a signal, the frequency of the tone must be within + or - 8 Hz, while the frequency of the tone corresponding to the event must be within + or - 30 Hz.

Continuity Test (co2):

A tone at 1780 Hz (see section 3.1.1.3 of [2]). When generated as a signal, the frequency of the tone must be within + or - 20 Hz, while the frequency of the tone corresponding to the event must be within + or - 30 Hz.

In continuity testing the tone corresponding to the signal at the originating gateway is referred to as the "go" tone, and the tone

corresponding to the event at that same gateway is referred to as the "return" or "check" tone.

Note that generation and notification of continuity tones are done as per continuity test requirements as defined in ITU-T Q.724 [3], as well as by Bellcore GR-317-CORE [2] specifications, i.e., the semantics of notification of the return tone is more than that the tone was received, but is an indication that the test has passed. Details are provided in the following paragraphs.

The continuity tones represented by col and co2 are used when the Call Agent wants to initiate a continuity test. There are two types of tests, single tone and dual tone; In the case of the dual tone, either tone can be sent and the opposite received depending on the trunk interconnections (4-wire or 2-wire) as indicated below:

Originating =====	Terminating =====
4w ----- 1780 Hz -----> <----- 2010 Hz -----	2w (transponder)
2w ----- 2010 Hz -----> <----- 1780 Hz -----	2w/4w (transponder)
4w ----- 2010 Hz -----> <----- 2010 Hz -----	4w (loopback)

The Call agent is expected to know, through provisioning information, which test should be applied to a given endpoint. As an example, for a 4-wire to 2-wire connection, the Call Agent might send a request like the following to an originating gateway:

```
RQNT 1234 ds/dsl-1/17@tgw2.example.net
X: AB123FE0
S: t/col
R: t/co2,t/oc,t/of
```

On a terminating side of a trunk, the call agent may request a continuity test connection (connection mode "conttest") to the terminating gateway as follows:

```
CRCX 3001 ds/dsl-2/4@tgw34.example.net
C: 3748ABC364
M: conttest
```

Alternatively, rather than using a connection mode, the "T/ct" signal can be used (see description of this signal further below):

```
RQNT 3001 ds/ds1-2/4@tgw34.example.net
X: 1233472
S: t/ct(in=col,out=co2,+)
```

The originating gateway would send the requested "go" tone, and would look for the appropriate "return tone". Once the return tone is received, the originating gateway removes the go tone and checks to see that the return tone has been removed within the specified performance limits (i.e., GR-246-CORE, T1.113.4, Annex B [23]). When it detects that the test is successful, the gateway will send a notification of the return tone event (Note that notification of the return tone event therefore must not be sent prior to detection of the removal of the return tone).

The "T/co1" and "T/co2" signals are TO signals so that an operation complete event will occur when the signal times out. If a timeout value other than the default is desired, the "to" parameter may be used (e.g., "S: T/co1(to=2000)").

If the gateway detects the failure of the continuity test prior to the timeout, an operation failure event will be generated. Otherwise, the failure of the continuity test is determined by the failure to receive the return tone event before the timeout occurs (operation complete event). As with TO signals in general, operation complete and operation fail events are parameterized with the name of the signal.

In the example above where the go tone is "col" and the return tone is "co2":

- \* A notification of the "co2" event indicates success (i.e., "O: t/co2").
- \* A notification of the operation failure event indicates failure prior to timeout (i.e., "O: t/of(t/co1)").
- \* A notification of the operation complete event indicates that the return tone was not received properly prior to the occurrence of the timeout (i.e., "O: t/oc(t/co2)").

On a terminating end of a trunk, either a "loopback" connection (single tone test) or "conttest" connection (dual tone test) is made (or alternatively the "T/lb" or "T/ct" signals are requested). It is up to the termination end to make sure that the return tone is removed as soon as the go tone disappears. The

Call Agent requests the removal of "contest" or "loopback" connections (or "T/lb" or "T/ct" signals) at a termination end when the results of the continuity test are obtained.

When "conttest" is used, the endpoint is provisioned as to which transponder test is being performed (2010 Hz received and 1780 Hz sent or vice versa). In the case of the corresponding "T/ct" signal, the Call Agent can specify which tone is received and sent as parameters.

Note that continuity tones in the trunk package are only ever sent to the telephony endpoint. For network-based continuity, there are continuity tones available in the RTP ("R") package. Although a transponder (dual tone) test can be done, a single tone test is generally sufficient in the case of continuity testing across an IP network.

Continuity Transponder(ct(in=<tone-in>,out=<tone-out>, <+ or ->):

This signal is used to provide transponder functionality independent of the connection mode, i.e., this is an alternative way to provide the same functionality as the "conttest" connection mode. The parameters can be provided in any order. The <tone-in> and <tone-out> parameters can have values "co1" or "co2", corresponding to the 2010 Hz and 1780 Hz tones associated with those symbols. If one of the tones is "co1", then the other must be "co2" and vice versa (i.e., <tone-in> and <tone-out> must have different values; if loopback is required, then the "lb" signal in this package or "loopback" connection mode should be used).

On detecting <tone-in>, <tone-out> will be generated in return. The tone corresponding to <tone-out> will continue to be generated until either:

- \* The signal is explicitly turned off (e.g., "S: t/ct(-)") or
- \* Removal of the <tone-in> tone is detected.

Note that while the signal is active (regardless of whether a tone is active or not), media from the endpoint will not be forwarded to or from the packet network (i.e., the continuity transponder signal must be explicitly turned off by the Call Agent in order to resume passing media between the packet network and the endpoint).

Loopback (lb):

This signal is used to provide loopback functionality independent of the connection mode, i.e., this is an alternative way to provide the same functionality as "loopback" connection mode.

Note that while the loop-back signal is active (regardless of whether a tone is active or not), media from the endpoint will not be forwarded to or from the packet network (i.e., the loopback signal must be explicitly turned off by the Call Agent in order to resume passing media between the packet network and the endpoint).

New Milliwatt Tone (nm):

1004 Hz tone - refer to [4] and section 8.2.5 of [5].

Newest Milliwatt Tone (mm):

1013.8 Hz - refer to [4].

Operation Complete (oc):

This is the standard definition of operation complete [1].

Operation Failure (of):

This is the standard definition of operation failure [1].

Old Milliwatt Tone (om):

1000 Hz tone - refer to [4] and section 8.2.5 of [5].

Permanent Signal Tone (pst):

In North America, this tone is applied to a busy line verify/operator interrupt under specific circumstances as described in [17].

Quiet Termination (qt):

Quiet Termination is used in a 102 trunk test. Reference section 6.20.5 [5] as well as [4].

Reorder Tone(ro):

This maps to congestion tone in the ITU-T E.182 specification. In North America, reorder tone is a combination of two AC tones with frequencies of 480 and 620 Hertz and levels of -24 dBm each, to give a combined level of -21 dBm. The cadence for reorder tone is 0.25 seconds on, followed by 0.25 seconds off, repeating continuously (until time-out). See GR-506-CORE [7], Section 17.2.7.

Special Information Tone(sit(#)):

As described in ITU-T E.180 [8], the special information tone consists of a tone period in which 3 tones are produced followed by a silent period of 1 second (total TO period of approximately 2 seconds). When used as a signal, it MUST be parameterized with a parameter value from 1 to 7, with the following meaning as defined in SR-2275, section 6.21.2 of [5].

sit(1)	RO'	reorder SIT, intra-LATA
sit(2)	RO"	reorder SIT, inter-LATA
sit(3)	NC'	no circuit SIT, intra-LATA
sit(4)	NC"	no circuit SIT, inter-LATA
sit(5)	IC	intercept SIT
sit(6)	VC	vacant code SIT
sit(7)	IO	ineffective other SIT

When requested as an event, the event MUST be parameterized with a decimal number from 1 to 7 to indicate which tone the gateway is required to detect. The resulting notification also includes the parameter. Other countries may have one or more special information tones with country specific definitions (refer to ITU-T E.180 supp. 2 [9]). In this case, special information tone 1 as defined in [9] is sit(1), special information tone 2 is sit(2) etc.

As an example, the Call Agent might make a request such as:

```
RQNT 1234 ds/ds1-1/17@tgw2.example.net
X: AB123FE0
R: t/sit(N)(2)
```

If the tone is detected, the resulting notification might appear as follows:

```
NTFY 3002 ds/ds1-3/6@gw-o.whatever.net MGCP 1.0
X: AB123FE0
O: t/sit(2)
```

Test Line (tl):

105 Test Line test progress tone (2225 Hz + or - 25 Hz at -10 dBm0). Refer to section 8.2.5 of [5].

Test Pattern (tp(###)):

The tp(###) signal inserts the pattern ### continuously into the channel until the timeout period expires. The parameter is provided as a decimal number from 0 to 255. If the parameter is omitted, the default value is decimal 95.

In RequestedEvents, the parameter MAY be supplied to indicate what pattern the Call Agent wishes the gateway to detect. If the parameter is omitted, the value 95 is assumed. The pattern MUST be returned in the ObservedEvent (even if the parameter was not requested).

A typical use for the test pattern signal is for the test line 108 (digital loopback) test (refer to section 8.2.5 of [5]). At the termination side of a trunk, the Call Agent would request a connection in "loopback" mode, which would do a digital loopback. On the origination side of the trunk, the Call Agent would request that the test pattern be injected into the digital channel, and would check to see that the pattern was returned within the timeout period. As an example, the Call Agent would make the following request on the origination side:

```
RQNT 1234 ds/dsl-1/17@tgw2.example.net
X: AB123FE0
S: t/tp
R: t/tp, t/oc, t/of
```

In this case the Call Agent will either receive:

- \* An ObservedEvent indicating that the test has passed (i.e., "O:t/op(95)") or
- \* An ObservedEvent indicating that the timeout occurred before the pattern was received (i.e., "O:t/oc(t/tp)"), indicating that the test failed. Of course an operation failure would indicate failure as well.

No Circuit (zz):

This is an alias for Special Information Tone 2, i.e., "sit(2)".

## 2.4. Line Package

Package Name: L  
Version: 1

Symbol	Definition	R	S	Duration
adsi(string)	ADSI Display		BR	
aw	Answer Tone	x	OO	
bz	Busy Tone		TO	30 sec.
ci(ti,nu,na)	Caller-id		BR	
dl	Dial Tone		TO	16 sec.
e	Error Tone	x	TO	2 sec.
hd	Off-hook Transition	S		
hf	Flash-hook	x		
ht	On Hold Tone		OO	
hu	On-hook Transition	S		
lsa	Line Side Answer Sup.		OO	
mwi	Message Waiting ind.		TO	16 sec.
nbz	Network busy	x	TO	infinite
oc	Operation Complete	x		
of	Operation Failure	x		
osi	Network Disconnect		TO	900 ms
ot	Off-hook Warning Tone		TO	infinite
p	Prompt Tone	x	BR	
rg	Ringing		TO	180 sec.
r0, r1, r2, r3, r4, r5, r6 or r7	Distinctive Ringing		TO	180 sec.
ro	Reorder Tone		TO	30 sec.
rs	Ringsplash		BR	
s(###)	Distinctive Tone Pattern	x	BR	
sit(#)	Special Information Tone		TO	2 sec. (see notes)
sl	Stutter Dial Tone		TO	16 sec.
v	Alerting Tone		OO	
vmwi	Visual Message Waiting Indicator		OO	
wt	Call Waiting Tone		TO	12 sec
wt1, wt2, wt3, wt4	Alternative Call Waiting Tones		TO	12 sec (see notes)
y	Recorder Warning Tone		TO	infinite
z	Calling Card Service Tone		BR	

New events added to this package from the previously unversioned package: "ht", "osi", and "lsa".

Changes in event types: signals "y", "z", changed from OO to TO and BR respectively. Ringing tones were extended to allow for a ring repetition signal parameter.

Note that default time-out values may be over-ridden by the Call Agent for any Time-Out signal defined in this package by a "to" signal parameter. Refer to section 2 of this document, as well as [1] for details.

The description of events and signals in the line package are as follows:

ADSI Display (adsi):

This signal is included here to maintain compatibility with the previous version of this package. The signal is not well-defined and its use is discouraged.

Answer Tone (aw):

This event is included here to maintain compatibility with the previous version of this package. The event is not well-defined and its use is discouraged.

Busy Tone (bz):

Refer to ITU-T E.180 [8]. In North America, station Busy is a combination of two AC tones with frequencies of 480 and 620 Hertz and levels of -24 dBm each, to give a combined level of -21 dBm. The cadence for Station Busy Tone is 0.5 seconds on followed by 0.5 seconds off, repeating. See GR-506-CORE [7], Section 17.2.6. It is considered an error to try and play busy tone on a phone that is on-hook and an error MUST consequently be returned when such attempts are made (error code 402 - phone on-hook).

Caller-id (ci(time, number, name)):

See GR-1188 [24], GR-30-CORE [14], and GR-31 [25]. For backwards compatibility, each of the three fields are optional, but each of the commas will always be included. In accordance with the general MGCP grammar, it is RECOMMENDED to always include all three fields - an empty quoted string can then be used in lieu of omitting a parameter:

The time parameter is coded as "MM/DD/HH/MM", where MM is a two-digit decimal value for a Month between 01 and 12, DD is a two-digit value for a Day between 01 and 31, and Hour and Minute are two-digit values coded according to military local time, e.g., 00 is midnight, 01 is 1 a.m., and 13 is 1 p.m. (Note: two digits MUST always be provided for each of the values of month, day, hour, minutes e.g., the month of January is indicated by the two digits "01" rather than just "1").

The number parameter is coded as an ASCII character string of decimal digits that identify the calling line number. White spaces are permitted if the string is quoted, but they will be ignored. If a quoted-string is provided, the string itself is UTF-8 encoded (RFC 2279) as usual for signal parameters.

The name parameter is coded as a string of ASCII characters that identify the calling line name. White spaces are permitted if the string is quoted. If a quoted-string is provided, the string itself is UTF-8 encoded (RFC 2279).

A "P" in the number or name field is used to indicate a private number or name, and an "O" is used to indicate an unavailable number or name. Other letters MAY be used to provide additional clarification as per provider or vendor specifications.

The following example illustrates the use of the caller-id signal:

```
S: 1/ci(09/14/17/26, "555 1212", "John Doe")
```

An example indicating that the name and number are private:

```
S: 1/ci(09/14/17/26,P,P)
```

#### Dial Tone (dl):

Refer to the ITU-T E.180 [8] specification. In North America, dial tone is a combination of two continuous AC tones with frequencies of 350 and 440 Hertz and levels of -13dBm each, to give a combined level of -10 dBm. See GR-506-CORE [7] - LSSGR: SIGNALING, Section 17.2.1. It is considered an error to try and play dial-tone on a phone that is on-hook and an error MUST consequently be returned when such attempts are made (error code 402 - phone on-hook).

#### Error Tone (e):

This tone is maintained for backwards compatibility. The tone is not well defined and its use is discouraged.

#### Off-hook Transition (hd):

See GR-506-CORE [7], Section 12. It is considered an error to try and request off-hook on a phone that is off-hook and an error MUST consequently be returned when such attempts are made (error code 401 - phone off-hook).

**Flash Hook (hf):**

See GR-506-CORE [7], Section 12. It is considered an error to try and request flash hook on a phone that is on-hook and an error MUST consequently be returned when such attempts are made (error code 402 - phone on-hook).

**Tone On Hold (ht):**

A tone used to reassure a calling subscriber who has been placed on "hold". Refer to ITU-T E.182 [10].

**On-hook Transition (hu):**

See GR-506-CORE [7], Section 12. The timing for the on-hook signal is for flash response enabled, unless provisioned otherwise. It is considered an error to try and request flash hook on a phone that is on-hook and an error MUST consequently be returned when such attempts are made (error code 402 - phone on-hook).

**Line Side Answer Supervision (lsa):**

This provides Reverse Loop Current Feed (RLCF) on the line (refer to GR-506-CORE [7]) and is a way of indicating that the called party has answered for some line-side equipment.

**Message Waiting Indicator (mwi):**

Message Waiting indicator tone uses the same frequencies and levels as dial tone (350 and 440 Hertz at -13dBm each), but with a cadence of 0.1 second on, 0.1 second off, repeated 10 times, followed by steady application of dial tone. See GR-506-CORE [7], Section 17.2.3. It is considered an error to try and play message-waiting indicator on a phone that is on-hook and an error MUST consequently be returned when such attempts are made (error code 402 - phone on-hook).

**Network Busy (nbz):**

This is included here to maintain compatibility with the previous version of this package. The "nbz" signal is an alias for re-order tone signal("ro"). Future Call Agent implementations that require a network busy signal should use the "ro" signal. It is also recommended that future Call Agents not request to be notified of the "nbz" event (a network busy event is generally not required in a line package; hence, "ro" is only a signal, not an event).

**Operation Complete (oc):**

This is the standard definition of operation complete [1].

**Operation Failure (of):**

This is the standard definition of operation failure [1].

**Network Disconnect (osi):**

Network Disconnect indicates that the far-end party has disconnected. The signal that is sent on the line is provisioned in the media gateway since it may vary from country to country. In North America, this signal is an open switch interval which results in a Loop Current Feed Open Signal (LCFO) being applied to the line (refer to GR-506-CORE [7], see also See GR-505-CORE [6], Section 4.5.2.1). The default time-out value for this signal is 900 ms.

**Off-hook Warning Tone (ot):**

Off-hook warning tone, also known as receiver Off-Hook Tone (ROH Tone). This is the irritating noise a telephone makes when it is not hung up correctly. In North America, ROH Tone is generated by combining four tones at frequencies of 1400 Hertz, 2060 Hertz, 2450 Hertz and 2600 Hertz, at a cadence of 0.1 second on, 0.1 second off, then repeating. GR-506-CORE [7], Section 17.2.8 contains details about required power levels. It is considered an error to try and play off-hook warning tone on a phone that is on-hook, and an error MUST consequently be returned when such attempts are made (error code 402 - phone on-hook).

**Prompt Tone (p):**

The definition of the prompt tone and its use may be found in requirement GR-220 [20]. The tone in GR-220 (requirement "R3-170" or GR-220) is a 300 ms burst of a 400 Hz tone.

**Ringing (rg):**

See GR-506-CORE [7], Section 14. The provisioning process may define the ringing cadence. The ringing signal may be parameterized with the signal parameter "rep" which specifies the maximum number of ringing cycles (repetitions) to apply. The value for "rep" is specified in decimal and can have any value from 1 to 255. The following will apply the ringing signal for up to 6 ringing cycles:

```
S: 1/rg(rep=6)
```

If the "rep" parameter is specified, the signal times-out when the number of repetitions are completed (i.e., an operation complete event can be requested and will occur at the end of the timeout/number of rings).

If the "rep" parameter is supplied, then any timeout ("to") value that is included will be ignored, i.e.:

```
S: 1/rg(rep=6,to=12000)
```

will be treated the same as the previous example where the parameter "to=12000" was not included. Of course, if the "to" parameter is included without the "rep", it will be acted upon i.e.:

```
S: 1/rg(to=12000)
```

will ring for 12 seconds.

It is considered an error to try and ring a phone that is off-hook and an error MUST consequently be returned when such attempts are made (error code 401 - phone off-hook).

Distinctive Ringing (r0, r1, r2, r3, r4, r5, r6 or r7):

See GR-506-CORE [7], Section 14. Default values for r1 to r5 are as defined for distinctive ringing pattern 1 to 5 in GR-506-CORE [7]. The default values for r0, r6 and r7 are normal ringing (i.e., the same cadence "rg"). The provisioning process may define the ringing cadence for each of these signals. The distinctive ringing signals may be parameterized with the signal parameter "rep" which specifies the maximum number of ringing cycles (repetitions) to apply. The value for "rep" is specified in decimal and can have any value from 1 to 255.

The following will apply the ringing signal for up to 6 ringing cycles:

```
S: 1/r1(rep=6)
```

If the "rep" parameter is specified, the signal times-out when the number of repetitions are completed (i.e., an operation complete event can be requested and will occur at the end of the timeout/number of rings)

If the "rep" parameter is supplied, then any timeout ("to") value that is included will be ignored, i.e.:

```
S: 1/r1(rep=6,to=12000)
```

will be treated the same as the previous example where the parameter "to=12000" was not included. Of course, if the "to" parameter is included without the "rep", it will be acted upon i.e.:

```
S: 1/r1(to=12000)
```

will ring for 12 seconds.

It is considered an error to try and ring a phone that is off-hook and an error MUST consequently be returned when such attempts are made (error code 401 - phone off-hook).

#### Reorder Tone (ro):

This maps to congestion tone in the ITU-T E.182 [10] specification. In North America, reorder tone is a combination of two AC tones with frequencies of 480 and 620 Hertz, and levels of -24 dBm each, to give a combined level of -21 dBm. The cadence for reorder tone is 0.25 seconds on, followed by 0.25 seconds off, repeating continuously.

#### Ringsplash (rs):

Also known as "Reminder ring", this tone 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 Forward sub-feature is active. In the US, it is defined to be a 0.5(-0,+0.1) second burst of power ringing (see [11]).

#### Distinctive Tone Pattern (s(###)):

This is used to signal or detect a tone pattern defined by the parameter where the parameter may have a value from 0 to 999. When specified as an event, the parameter MUST be included. The parameter will also be included when the event is reported. This event (the definition of tones associated with each parameter value) requires special provisioning in the Call Agent and gateway to insure interoperability. This signal is included here to maintain compatibility with the previous version of this package.

#### Special Information Tone(sit(#)):

As described in ITU-T E.180 [8], the special information tone consists of a tone period in which 3 tones are produced, followed by a silent period of 1 second (total TO period of approximately 2 seconds). It MAY be parameterized with a parameter value from 1 to 7, with the following meaning as defined in SR-2275, section 6.21.2 [5]:

sit(1)	RO'	reorder SIT, intra-LATA
sit(2)	RO"	reorder SIT, inter-LATA
sit(3)	NC'	no circuit SIT, intra-LATA
sit(4)	NC"	no circuit SIT, inter-LATA
sit(5)	IC	intercept SIT
sit(6)	VC	vacant code SIT
sit(7)	IO	ineffective other SIT

If the parameter is left out, the NC' SIT tone that corresponds to the signal "L/sit(3)" is assumed.

Other countries may have one or more special information tones with country specific definitions (refer to ITU-T E.180 supp. 2 [9]). In this case, special information tone 1 as defined in [9] is sit(1), special information tone 2 is sit(2) etc.

#### Stutter Dial Tone (sl):

Stutter Dial Tone (also called Recall Dial Tone in GR-506-CORE [7] and "special dial tone" in ITU-T E.182 [10]) is used to confirm some action and request additional input from the user. An example application is to cancel call-waiting, prior to entering a destination address.

The stutter dial tone signal may be parameterized with the signal parameter "del", which will specify a delay in milliseconds to apply between the confirmation tone and the dial tone. The parameter can have any value from 0 to 10000 ms, rounded to the nearest non-zero value divisible by 100 (i.e., tenth of a second). The following will apply stutter dial tone with a delay of 1.5 seconds between the confirmation tone and the dial tone:

```
S: 1/sl(del=1500)
```

It is considered an error to try and play stutter dial tone on a phone that is on-hook and an error MUST consequently be returned when such attempts are made (error code 402 - phone on-hook).

#### Alerting Tone (v):

A 440 Hz Tone of a 2 second duration, followed by a 1/2 second of tone every 10 seconds. This event is included for backwards compatibility with the previous version of the package.

#### Visual Message Waiting Indicator (vmwi):

The transmission of the VMWI messages will conform to the requirements in [13] and the CPE guidelines in [12]. Refer also to section 6.6 of GR-30-CORE [14]. VMWI messages will only be sent from the gateway to the attached equipment when the line is idle. If new messages arrive while the line is busy, the VMWI indicator message will be delayed until the line goes back to the idle state. After the gateway restarts, the state of the signal will be "off", and hence the Call Agent MUST refresh the CPE's visual indicator if it is supposed to be "on".

**Alternative Call Waiting Tones (wt, wt1, ..., wt4):**

Refer to ITU-T E.180 [8]. For North American tone definitions, refer to GR-506-CORE [7], Section 14.2. "wt" and "wt1" are both aliases for the default Call Waiting tone, which in North America, is a 440-Hz tone applied for 300 plus or minus 50 ms. The tone is then repeated once after 10 seconds.

These signals are timeout signals with a default timeout value of 12 seconds, which allows the tone to be played twice with a single request (refer to GR-571-CORE [16]). However, there are cases (Requirement R3-73 of GR-575-CORE [18]), in which only a single tone is required. In that case, the Call Agent may make the request with a shorter timeout period to eliminate the second tone (e.g., "S: wt(to=2000)" - which stops the signal after 2 seconds so that the second tone will not occur).

Signals wt2, wt3 and wt4 are alternates that are used for distinctive call-waiting tone patterns (refer to GR-506-CORE, Section 14.2 [7]). It is considered an error to try and apply call-waiting tone on a phone that is on-hook and an error MUST consequently be returned when such attempts are made (error code 402 - phone on-hook).

**Recorder Warning Tone(y):**

Refer to ITU-T E.180 [8] - also Bellcore document SR-2275 [5] section 6.20. When recording equipment is used, this tone is connected to the line to inform the distant party that the conversation is being recorded - typical value used is a 1400 Hz Tone of a 0.5 second duration every 15 seconds.

**Calling Card Service Tone(z):**

This tone is used to inform the customer that credit card information must be keyed in. Typically, it consists of 60 ms of 941 + 1477 Hz (the DTMF #digit) and 940 ms of 350 + 440 Hz (dial tone), decaying exponentially with a time constant of 200 ms. Refer to Bellcore document SR-2275 [5], section 6.20.

## 2.5. Handset Emulation Package

Package Name: H

Version: 1

Symbol	Definition	R	S	Duration
adsi(string)	ADSI Display	x	BR	
aw	Answer Tone	x	OO	
bz	Busy Tone	x	TO	30 sec.
ci(ti,nu,na)	Caller-id	x	BR	
dl	Dial Tone	x	TO	16 sec.
e	Error Tone	x	TO	2 sec.
hd	Off-hook Transition	S	BR	
hu	On-hook Transition	S	BR	
hf	Flash Hook	x	BR	
ht	Tone On Hold	x	OO	
lsa	Line Side Answer Sup.	x	OO	
mwi	Message Waiting Ind.	x	TO	16 sec.
nbz	Network Busy	x	TO	infinite
oc	Operation Complete	x		
ot	Off-hook Warning Tone	x	TO	infinite
of	Operation Failure	x		
osi	Network Disconnect	x	TO	900 ms
p	Prompt Tone	x	BR	
rg	Ringing	x	TO	180 sec.
r0, r1, r2, r3, r4, r5, r6 or r7	Distinctive Ringing	x	TO	180 sec.
ro	Reorder Tone	x	TO	30 sec.
rs	Ringsplash	x	BR	
s(###)	Distinctive Tone Pattern	x	BR	
sit(#)	Sit Tone	x	TO	2 sec.
sl	Stutter Dial Tone	x	TO	16 sec.
v	Alerting Tone	x	OO	
vmwi	Vis. Message Waiting Ind.	x	OO	
wt	Call Waiting tone	x	TO	12 sec.
wt1, wt2, wt3, wt4	Alternative Call Waiting Tones	x	TO	12 sec (see notes)
y	Recorder Warning Tone	x	TO	infinite
z	Calling Card Serv. Tone	x	BR	

The handset emulation package is similar to the line package except that events such as "off-hook" can be signaled as well as detected.

Changes from the original package - are the same changes as were made for the line package, plus "hu" and "hd" signal types were changed from OO to BR.

Event definitions are the same as for the line package with the following exceptions:

ASDI:

When requested as an event by the Call Agent, the event is not parameterized. However, the parameter is included when the event is reported.

Caller-id:

When requested as an event by the Call Agent, the event MUST not be parameterized. However, parameters are included when the event is reported i.e.:

```
O: l/ci(09/14/17/26,"555 1212","John Doe")
```

Line Side Answer Supervision:

When requested as an event by the Call Agent, it indicates when the reverse loop current feed (RLCF) was turned on and off. The event is not parameterized when it is requested. However, a parameter is included when it is reported i.e.:

```
O: l/lsa(+) to indicate RLCF was turned on  
O: l/lsa(-) to indicate RLCF was turned off
```

Ringing (rg):

When requested as an event, the Call Agent may optionally include the rep parameter indicating a request to report after some number of rings e.g.:

```
RQNT 1234 aaln/1@rgw2.example.net  
X: AB123FE0  
R: h/rg(N)(rep=3)
```

The resulting notification after the number of rings is detected includes the parameter again:

```
NTFY 3002 ds/dsl-3/6@gw-o.whatever.net MGCP 1.0  
X: AB123FE0  
O: h/rg(rep=3)
```

If the parameter is not included in the request, it is also not included in the report. In that case, the event is reported as soon as ringing is detected.

**Distinctive Ringing (r0, r1, r2, r3, r4, r5, r6 or r7):**

As with the "rg" event, if the "rep" parameter is included when one of these is requested as an event, it is also reported. If it is not requested with the parameter, then the parameter is also not included in the report. In that case, the event is reported as soon as ringing with the requested cadence is detected.

**Stutter Dial Tone (sl):**

Stutter Dial Tone MUST not be parameterized when requested as an event. However, the "del" parameter is reported.

```
RQNT 1234 aaln/1@rgw2.example.net
X: AB123FE0
R: h/sl
```

The resulting notification indicates the delay between the confirmation tone and the dial tone:

```
NTFY 3002 ds/dsl-3/6@gw-o.whatever.net MGCP 1.0
X: AB123FE0
O: h/sl(del=1500)
```

As with the signal, the report indicates the delay rounded to the nearest 100 ms.

**Visual Message Waiting:**

When requested as an event by the Call Agent, it indicates when the visual message waiting indicator was turned on and off. The event is not parameterized when it is requested. However, a parameter is included when it is reported i.e.:

```
O: l/vmwi(+) to indicate message waiting turned on
O: l/vmwi(-) to indicate message waiting turned off
```

**Note that:**

- \* All TO signals in the handset package can include a "to" parameter when requested as a signal.
- \* However, requests to be notified about these events MUST NOT include the "to" parameter, i.e., the "to" parameter is not valid in RequestedEvents.

## 2.6. Supplementary Services Tone Package

Package Name: SST

Version: 0

Symbol	Definition	R	S Duration
cd	Conference Depart		BR
cj	Conference Join		BR
cm	Comfort Tone		TO infinite
cw	Caller Waiting Tone		TO 30 sec.
ht	On Hold Tone		OO
ni	Negative Indication		TO infinite
nu	Number Unobtainable		TO infinite
oc	Operation Complete	x	
of	Operation Failure	x	
pr	Pay Phone Recognition		BR
pt	Pay Tone		BR

Note that default time-out values may be over-ridden by the Call Agent for any Time-Out signal defined in this package by a "to" signal parameter. Refer to section 2 of this document, as well as [1] for details.

The events in this package are defined as follows:

## Conference Depart(cd):

Tone used to indicate that a participant has left a conference call. The tone characteristics are left to the specific gateway implementation.

## Conference Join (cj):

Tone used to indicate that a party has joined a conference call. The tone characteristics are left to the specific gateway implementation.

## Comfort Tone (cm):

Comfort Tone is used to indicate that the call is being processed and that the caller should wait. Refer to ITU-T E.182 [10].

## Caller Waiting Tone (cw):

Not to be confused with a call-waiting tone, this is a tone advising a caller that a called station, though busy, has a call waiting service active. Refer to ITU-T E.182 [10].

**Tone on-hold (ht):**

A tone used to reassure a calling subscriber who has been placed on "hold". Refer to ITU-T E.182 [10].

**Negative Indication (ni):**

A tone advising a subscriber that the request for service cannot be accepted. Refer to ITU-T E.182 [10]. For North America, this maps to the re-order tone (see GR-506-CORE [7], Section 17.2.7).

**Number Unobtainable Tone (nu):**

Refer to ITU-T E.180, supplement 2 [9]. This is also referred to as "vacant tone" and maps to a "re-order tone" in North America (see GR-506-CORE [7], Section 17.2.7).

**Operation Complete (oc):**

The standard definition of operation complete [1].

**Operation Failure (of):**

The standard definition of operation failure [1].

**Pay Phone Recognition (pr):**

A tone advising an operator that the endpoint is identified as a payphone. Refer to ITU-T E.182 [10].

**Pay Tone (pt):**

A tone indicating that payment is required. Refer to ITU-T E.182 [10].

## 2.7. Digit Map Extension

Package Name: dm1 ("dm" followed by the number "1")

Version: 0

Extension Digit Map Letters: P

This package defines an Extension Digit Map Letter that is used to override the shortest possible match behavior for a given entry in a digit map (see [1]). The letter "P" (for partial match override), at the end of a digit map entry, instructs the gateway to only consider that entry a match if the current dial string does not partially match another entry. For example, given the digit map

```
([3-7]11|123xxxxxx|[1-7]xxxxxP|8xxxP)
```

and a current dial string of "1234567", we would not consider this a match (as the rules in [1] would otherwise imply); however a current dial string of "411" would be considered a match as usual. A current dial string of "8234" would be considered a match since there is no other partial match.

Note that the digit map letter "P" is not an event, but simply a syntactic and semantic digit map extension. Thus, the "P" is not included in the list of requested or observed events.

Support for this package is strongly RECOMMENDED.

## 2.8. Signal List Package

Package Name: SL

Version: 0

Symbol	Definition	R	S	Duration
oc	Operation Complete	x		
of	Operation Failure	x		
s(list)	Signal List		TO	variable

Operation Complete (oc):

This is the standard definition of operation complete from [1].

Operation Failure (of):

This is the standard definition of operation failure from [1].

Signal List(s(<list>)):

The <list> contains a comma-separated list of signals to be played out. Each of the signals in <list> MUST be either of type BR or type TO. Semantically, the signal list is still treated as a single parameterized signal of type Time-Out though. The signals in the list are played to completion one after the other in the left to right order specified. The package for each signal in the list must be specified. For example, to play out the DTMF digits 123456:

```
S: sl/s(d/1,d/2,d/3,d/4,d/5,d/6)
```

This will result in the DTMF digits 1, 2, 3, 4, 5 and 6 being played out in order.

It is illegal to include an OO signal as one of the signals in the list or to request recursive definitions (signal lists within signal lists). If this or any other unsupported signal is included, error code 538 (event/signal parameter error) MUST be returned by the gateway.

Note that as the gateway plays the ordered list of signals, if it encounters a TO signal with an infinite timeout, it will continue to play that signal until the Signal List signal is stopped (i.e., other signals later in the list will never be played).

If the operation complete ("oc") event is requested, it will be detected once, when the last signal in the list has been played out (regardless of whether there are any TO signals in the list). The operation complete event will only report the signal list name itself, i.e., without the parameters supplied as in:

```
O: sl/oc(sl/s)
```

Should any of the signals in the signal list result in an error, an operation failure event for the Signal List signal MUST be generated. Only the signal list name will be included, thus it is not possible to determine which of the signals in the signal list actually failed.

Note that if an event occurs while the "SL/S" signal is playing, the "SL/S" signal is stopped in the following manner:

- \* If the signal in the list that was playing at the time the event occurred is of type BR, then the BR signal will be played to completion and no other signals in the list will be played.
- \* If the signal in the list that was playing at the time the event occurred is of type TO, then the TO signal will stop immediately and no other signals in the list will be played.

## 2.9. Media Format Parameter Package

Package Name: FM

Version: 0

This package provides support for the media format parameter Local Connection Option (LCO). The media format parameter LCO is similar to the "fmt" attribute in SDP [15] and is applicable to all of the same media formats that the corresponding SDP fmt attribute could be used with (i.e., media format parameters for any media format MIME type). The media format parameter is encoded as the keyword "fmt" or "o-fmt", followed by a colon and a quoted string beginning with the media format name (MIME subtype only) followed by a space, followed by the media format parameters associated with that media format. For simplicity, we will use the terms "codec" and "media

format" interchangeably in the following. Multiple formats may be indicated by either repeating the "fmt" local connection option multiple times, such as:

```
L:a:codec1;codec2, fmt:"codec1 formatX", fmt:"codec2 formatY"
```

or alternatively by having a single "fmt" keyword followed by a colon, and a semi-colon separated list of quoted strings for each media format parameter, as in:

```
L:a:codec1;codec2, fmt:"codec1 formatX";"codec2 formatY"
```

The two formats may be mixed.

If it is possible for the same codec to be requested with and without the special "fmt" format, the following could result:

```
L:a:codec1;codec1, fmt:"codec1 formatX"
```

However, it would not be clear if the fmt parameter was to be applied to the first or the second occurrence of the codec. The problem with that is, that codec ordering is important (i.e., codecs are listed in preferred order), and the above syntax does not provide a way to indicate if "formatX" is preferred (i.e., associated with the first "codec1") or not (i.e., associated with the second "codec1"). In order to resolve this dilemma, when the same codec is requested with multiple formats, the codec name in the "fmt" format string is followed by a colon and an <order>, where <order> is a number from one to N for N occurrences of the same codec in the codec list i.e.:

```
L:a:codec1;codec1, fmt:"codec1:2 formatX"
```

indicates that "formatX" is associated with the second instance of "codec1" in the "a:codec1;codec1" list. If an invalid instance number is supplied (e.g., instance 3 where there are only two instances), then error code 524 - inconsistency in local connection options will be returned.

Pre-pending "fmt" with the string "o-" (i.e., "o-fmt") indicates that the format is optional. In that case, the gateway may decide not to use the fmt parameter specified, or only use it in part.

If the "fmt" in an LCO is not optional (i.e., does not have "o-" in front of it), and the LCO value is either not recognized or not supported, then the associated codec is considered "not supported".

When auditing capabilities, the "fntp" local connection option MUST be returned with a semi-colon separated list of supported formats and/or multiple independent "fntp" parameters as in:

```
A: a:telephone-event, fntp:"telephone-event 0-15,32-35",...
```

```
A: a:PCMU;G729, fntp:"PCMU foo";"PCMU bar", fntp:"G729 foobar",...
```

One example uses the media format parameter LCO in conjunction with the media format "telephone-event", as defined in RFC 2833 [33]. If the media format "telephone-event" is used without the "fntp" media format parameter, the DTMF digits (telephone events 0-15 from RFC 2833 [33]) are assumed - such practice is however discouraged. On the other hand, the media format parameter LCO MAY be used to specify the exact set of events that are being requested via RFC 2833 [33]. Example:

```
L: a:PCMU;telephone-event, fntp:"telephone-event 16"
```

indicates that if telephone events are supported at all, then this request is specifically for event 16.

In another case, the Call Agent may indicate that some format parameters are "required", while others are optional. In the example below, telephone events 0-15 are a "must", while telephone events 16, 70 and 71 are optional.

```
L: a:PCMU;telephone-event, o-fntp:"telephone-event 16,70,71",  
fntp:"telephone-event 0-15"
```

If the gateway cannot support telephone events 0-15, it MUST NOT include the "telephone-event" media format in the SDP in its response. On the other hand, if it can support those telephone events, it SHOULD indicate support for those events, as well as any of the events 16, 70 and 71 that it supports.

If a request is made to audit the capabilities of an endpoint, and the endpoint supports the "telephone event" media format with events "0-16", then the audit would include the following:

```
A: a:telephone-event, fntp: "telephone-event 0-16"
```

Another example is the use of redundancy with RFC 2198 [32]. Again, the format of the fntp string is similar to that used in the SDP except that the literal string ("red" in this case) is used rather than the payload type:

```
L: a:G729;pcmu;red, fntp:"red pcmu/g729"
```

The corresponding media description in the SDP as part of the connection request acknowledgment might look like:

```
m=audio 12345 RTP/AVP 98 18 0
a=rtpmap:98 red/8000/1
a=fmtp:98 0/18
```

If we combine both telephone events and redundancy, an example local connection option might look as follows (carriage return added for formatting reasons here):

```
L: a:G729;pcmu:red;telephone-event,fmtp:"red pcmu/g729",
      fmtp: "telephone-event 16"
```

Note that we again specify the literal string for the encoding method rather than its payload type. This is a general principle that should be used with this LocalConnectionOption.

The corresponding SDP might appear as follows:

```
m=audio 12345 RTP/AVP 97 98 18 0
a=rtpmap:97 red/8000/1
a=fmtp:97 0/18
a=rtpmap:98 telephone event
a=fmtp:98 16
```

Note that the fmtp LCO may be used in any situation where the corresponding SDP attribute may be used. An example of a local connection option that involves a media type other than audio and a "foobar" fmtp parameter:

```
L: a:image/tiff, fmtp:"tiff foobar"
```

Note that normally local connection options that are associated with a package should have the package prefix included as per the package extension rules in [1]. The "fmtp" and "o-fmtp" LCO in the "FM" package are an exception. The package prefix is not included in the case of the "fmtp" and "o-fmtp" local connection options because they were created before the extension rules in [1] were defined.

These two LocalConnectionOptions have been registered with IANA.

## 2.10. RTP Package

Package Name: R  
Version: 1

Symbol	Definition	R	S Duration
col	Continuity Tone (single or return tone)	C	TO,C 3 sec.
co2	Continuity Test (go tone, in dual tone procedures)	C	TO,C 3 sec.
iu(..)	ICMP Unreachable Received	C	
ji(..)	Jitter Buffer Size Changed	C	
ma	Media Start	C	
oc	Operation Complete	x	
of	Operation Failure	x	
pl(..)	Packet Loss Exceeded	C	
qa	Quality Alert	C	
rto(..)	RTP/RTCP Timeout	C	
sr	Sampling Rate Changed	C	
uc	Used Codec Changed	C	

Changes in event types: "col" and "co2" signals changed from OO to TO.

New events added to this package from the previously unversioned package: "iu", "rto", "ma".

Note that default time-out values may be over-riden by the Call Agent for any Time-Out signal defined in this package by a "to" signal parameter. Refer to section 2 of this document, as well as [1] for details.

The events in this package all refer to media streams (connections), i.e., they cannot be detected on an endpoint. Furthermore, with the exception of the "iu" event, which is defined for any type of media, all other events in this package are defined for RTP media streams only (i.e., if they are used on connections that do not use RTP, the behavior is not defined).

Signals requested (e.g., "col" and "co2") must indicate the connection ID (e.g., "S: r/col@connectionID"). An event may be requested for all existing connections using the "\*" wildcard for the connectionID as described in [1].

## Example:

```
R: r/uc@*      (request to detect uc on all connections) or
R: r/uc@connectionID  (request to detect uc only on a specific
connection)
```

An event detected on a connection will include the connectionID,  
e.g.:

```
O: r/uc@connectionID(15)
```

## Continuity tones (col and co2):

These are the same as the events defined in the Trunk package, except in this case, they are only played over a network connection and the connectionID MUST be supplied (e.g., "s:r/col@connectionID"). They can be used in conjunction with the Network LoopBack (netwloop) or Network Continuity Test (netwtest) modes to test the continuity of an RTP circuit. However, in the case of testing IP continuity, a one-tone test is sufficient i.e., generating and detecting "col" at one end, with connection mode in network loopback mode at the other end. Note that the test can also be done using telephone events rather than tones, i.e., event 167 in RFC 2833 [33] corresponds to "col". In this case, connection requests are made with local connection options such as:

```
L: a:PCMU;telephone-event,fmtp:"telephone-event 167"
```

in order to request support for telephone event 167. If both ends support the event, then the network loopback proceeds as usual, except that telephone events corresponding to the col tone are sent, as opposed to the col tone itself.

## ICMP Unreachable Received (iu):

This event indicates that some number of ICMP unreachable packets [19] was received for this connection since an RQNT was received requesting this event. This notification indicates that packets that were sent by the gateway on this connection either did not arrive at their destination or were not accepted (e.g., the port was closed). When this event is requested, a single parameter with a decimal number from 1 to 255 may be included to indicate the number of ICMP un-reachable packets that must occur before the event is notified. If no parameter is supplied, with the request then a default value of 3 is assumed. This is a one-shot event in that once the event occurs, a further request is required in order to re-initiate counting.

The observed event is parameterized with two parameters:

- \* The first parameter is the number of ICMP unreachable packets received (i.e., the same value that was included in the request - or the value 3, if the requested event was not parameterized)
- \* The second parameter is the error code indicated in the ICMP unreachable packet, e.g.:
  - 0 = net unreachable;
  - 1 = host unreachable;
  - 2 = protocol unreachable;
  - 3 = port unreachable;
  - 4 = fragmentation needed and DF set;
  - 5 = source route failed.
  - etc.

An example of a request might be as follows:

```
RQNT 2001 ds/dsl-3/6@gw-o.whatever.net MGCP 1.0
X: 0123456789B0
R: r/iu@364823(N)(5)
```

In this case, a notify will occur if 5 ICMP port unreachable packets are received as a result of RTP and/or RTCP packets being sent from this gateway on the connection with connection ID 364823.

The resulting NOTIFY with observed events might be as follows:

```
NTFY 3002 ds/dsl-3/6@gw-o.whatever.net MGCP 1.0
X: 0123456789B0
O: r/iu@364823(5,3)
```

The first parameter indicates 5 ICMP unreachable packets were received since the RQNT with this request was sent. The second parameter ("3") specifies the reason, which in this case, is "port unreachable".

**Jitter Buffer Size Changed (ji):**

This event is only included here to maintain compatibility with the previous version of this package. This event is used to indicate that the gateway has made an adjustment to the depth of the jitter buffer. The syntax for requesting notification is "ji", which tells the media gateway that the controller wants notification of any jitter buffer size changes. The syntax for notification from the media gateway to the controller is "JI(####)", where the #### is a decimal number from 1 to 65536, indicating the new size of the jitter buffer in milliseconds.

**Media Start (ma):**

The media start event occurs on a connection when the first valid RTP media packet is received on the connection. This event can be used to synchronize a local signal, e.g., ringback, with the arrival of media from the other party.

The event is detected on a connection. If no connection is specified, the event applies to all connections for the endpoint, regardless of when the connections are created (i.e., if a connection is not specified, the event will occur when the first valid RTP packet arrives on any one of the connections on that endpoint).

**Operation complete (oc):**

This is the standard definition of operation complete [1].

**Operation failure (of):**

This is the standard definition of operation failure [1].

**Packet Loss Exceeded (pl):**

Packet loss rate exceeds the threshold of the specified decimal number (with a range of 1 to 100,000) of packets per 100,000 packets, where the packet loss number is indicated in parenthesis. For example, PL(10) is a drop rate of 10 in 100,000 packets. This event is requested with a parameter indicating at what packet loss rate the Call Agent wishes to be reported. If the packet loss exceeds that value, the event is reported with that same parameter. The event is only reported once when the packet loss threshold is exceeded. Once reported, a following request will re-initiate packet loss measurements and report when the threshold is exceeded again.

**Quality alert (qa):**

The packet loss rate or the combination of delay and jitter exceeding a quality threshold. The quality thresholds for delay, jitter and packet loss rate are provisioned values.

RTP/RTCP Timeout (rto(<timeout>,st=<start-time>)):

This event indicates that neither RTP nor RTCP packets have been received on this connection for a period of time equal to the <timeout> value (in seconds). The timeout value can be supplied as a decimal number from 1 to 65535 in the parameter when the request is made. The <timeout> parameter will be supplied in ObservedEvents when the event is reported - it then simply repeats the value used. If an RTP or RTCP packet is received before the timer expires, then the timer is reset and re-started. The event will only be generated if the timer expires without an RTP or RTCP packet arriving on the specified connection during the specified period of time. Note that if the event is requested without the <timeout> parameter, then a default timeout of 60 seconds is assumed. The <timeout> value will still be reported in ObservedEvents, even if no timeout value was indicated in the request (the default value will be indicated in that case). This is a one-shot event in that once the event occurs, a further request is required in order to re-initialize the timer.

Another optional <start-time> parameter may also be included. This is used to indicate when the timer starts. It can have one of the following values:

- \* "im" for immediate i.e., the timer starts as soon as the request is received. This is the default.
- \* "ra" to indicate that the timer should start only after an RTCP packet has been received from the other end (i.e., the timer will be initiated when the first RTCP packet is received after the request is made). Note that in the case where the other end does not support RTCP, the timer will never be initiated.

Note that either the <timeout> or <start-time> may be included in the request, but only the <timeout> value is included in the report.

An example of a request might be as follows:

```
RQNT 2001 ds/dsl-3/6@gw-o.whatever.net MGCP 1.0
X: 0123456789B0
R: r/rto@364823(N)(120,st=im)
```

In this case, a notify will occur if there is a period of time when no RTP or RTCP packets have been received on connection 364823 for 120 seconds.

The resulting NTFY with observed events would be as follows:

```
NTFY 3002 ds/ds1-3/6@gw-o.whatever.net MGCP 1.0
X: 0123456789B0
O: r/rto@364823(120)
```

Sampling Rate Changed (sr):

This event is only included here to maintain compatibility with the previous version of this package. This event indicates that the packetization period changed to some decimal number in milliseconds enclosed in parenthesis, as in SR(20).

Used Codec Changed (uc):

This event is only included here to maintain compatibility with the previous version of this package. This event is requested without a parameter, but when reported, the hexadecimal payload type is enclosed in parenthesis, as in UC(8), to indicate the codec was changed to PCM A-law. Codec Numbers are specified in RFC 3551 [26], or in a new definition of the audio profiles for RTP that replaces this RFC.

## 2.11. Resource Reservation Package

Package Name: RES  
Version: 0

### 2.11.1. Description

The "RES" package provides local connection option support for resource reservations as well as an event to indicate reservation loss.

A number of LocalConnectionOption parameters are used in doing resource reservations: "reservation request", "reservation direction", "reservation confirmation" and "resource sharing".

Reservation Request LocalConnectionOption: The gateways can be instructed to perform a reservation on a given connection using RSVP. When a reservation is needed, the Call Agent will specify the reservation profile that should be used, which is either "controlled load" or "guaranteed service". The absence of reservation can be indicated by asking for the "best effort" service, which is the default value for this parameter.

Whether or not RSVP will be done is dependent on whether the reservation request LocalConnectionOption parameter has been included in a connection request for this connection (with either "controlled load" or "guaranteed service" indicated). If a modify connection

(MDCX) request requires a change in the reservation and the "reservation request" parameter is not included in the LocalConnectionOptions, but was included in the LocalConnectionOptions for a previous connection request for that connection, then the "reservation request" value defaults to its previously saved value for that connection. If a modify connection (MDCX) request explicitly contains a "reservation request", indicating a request for "best effort" for a connection that has an existing reservation, the existing reservation will be torn down.

#### Reservation Direction LocalConnectionOption:

When reservation has been requested on a connection, the gateway will examine the reservation direction LocalConnectionOption parameter to determine the direction that the reservations require and do the following:

- \* Start emitting RSVP "PATH" messages if the reservation direction LocalConnectionOptions parameter specified "send-only" or "send-receive".
- \* Start emitting RSVP "RESV" messages as soon as it receives "PATH" messages if the reservation direction parameter specified "receive-only" or "send-receive".

If an RSVP reservation is requested, but the reservation direction LocalConnectionOption parameter is missing, the reservation direction defaults to the previously saved value of the reservation direction parameter for that connection. If there was no previous reservation direction parameter for that connection, the value is deduced from the connection mode. That is:

- \* Start emitting RSVP "PATH" messages if the connection is in "send-only", "send-receive", "conference", "network loop back" or "network continuity test" mode (if a remote connection descriptor has been received).
- \* Start emitting RSVP "RESV" messages as soon as it receives "PATH" messages if the connection is in "receive-only", "send-receive", "conference", "network loop back" or "network continuity test" mode.

#### Reservation Confirmation LocalConnectionOption:

Another LocalConnectionOption parameter for RSVP reservations is the reservation confirmation parameter, which determines what the resource reservation pre-condition (see [1]) is for acknowledging a successful connection request:

- \* If the reservation confirmation parameter is set to "none", the gateway will "Ack" the connection request without waiting for reservation completion. This is the default behavior.
- \* If the "reservation confirmation" parameter is set to "send-only", the gateway will "Ack" when the PATH message has been sent and the corresponding RESV is received to indicate successful reservation in the send direction.
- \* If the "reservation confirmation" parameter is set to "receive-only", the gateway will "Ack" when reservation confirm for a reservation has been received.
- \* If the reservation confirmation parameter is set to "send-receive", the gateway will "Ack" only after the PATH message has been sent and the corresponding RESV has been received for send direction, and reservation confirm has been received for the receive direction.

Note that:

Values "receive-only" and "send-receive" are triggers for the gateway to request reservation confirm (RESVCONF) when it sends out the RESV.

Pre-conditions SHOULD only be added for the direction(s) for which resource reservations have been requested. If a direction is added as a pre-condition, and that direction was not requested in the resource reservation, the direction MUST simply be ignored as a pre-condition.

In this approach, resource reservation success is the pre-condition to final acknowledgement of the connection request. If the reservation fails, the connection request also fails (error code 404 - insufficient bandwidth) - as will any other part of the transaction, e.g., a notification request included as part of the connection request. A typical example of this would be a request to ring the phone and look for off-hook, included with the connection request. If the reservation fails, the phone will not ring. Similarly, if the phone is already off-hook, the command fails and there will be no resource reservation.

A provisional response SHOULD be provided if confirmation is expected to occur outside the normal retry timers and in fact a provisional response MUST be provided if the reservation confirmation parameter has value "send-receive" (without a provisional response, SDP information cannot be returned until the

final "Ack" which will not occur until the reservation is complete. This can result in a deadlock since the SDP information typically needs to be passed to the other end in order for it to initiate the RSVP PATH message in the other direction). The SDP information and connectionID MUST be included in both the provisional response and the final response. Note that in order to ensure rapid detection of a lost final response, final responses issued after provisional responses for a transaction SHALL be acknowledged, i.e., they SHALL include an empty "ResponseAck" parameter in the final response (see [1]).

If the transaction time is outside the expected bounds (time T-HIST - see the section on provisional responses in [1]), error code 406 (transaction timeout) SHOULD be returned.

Also note that if the reservation confirmation parameter is omitted, the value of the reservation confirmation parameter defaults to its previously saved value. If there is no previously saved value for the reservation confirmation parameter, or the reservation confirmation parameter has the value "none", then successful resource reservation is not a pre-condition to providing an acknowledgement to the connection request (i.e., the gateway can "Ack" right away without waiting for the reservation to complete and a provisional response will not be necessary).

#### Resource Sharing LocalConnectionOption:

It may be possible to share network resources across multiple connections. An example is a call-waiting scenario, where only one connection will ever be active at a time. In a 3-way calling scenario with a similar set of connections, sharing is not possible. Only the Call Agent knows what may be possible, depending on the feature that is being invoked.

In order to allow the Call Agent to indicate that sharing is possible, a resource sharing LocalConnectionOption parameter is introduced. This parameter can have one of the following values:

- \* A value "\$" can be specified where \$ refers to "this connection". This value is used when doing a create connection and indicates the intent to share resources with this connection.
- \* A connection ID can be specified which indicates that this is a request to share resources with the connection having this connection ID (allowing multiple connections to share resources with the connection indicated).

- \* The value can be empty, which indicates a request to no longer share the resources of this connection with other connections.

In the case of a CRCX, the default value for the resource sharing local connection option is empty, and for an MDCX, the default value is its current value.

The RSVP filters will be deduced from the characteristics of the connection. The RSVP resource profiles will be deduced from the connection's bandwidth and packetization period.

Note that if RSVP is used with PacketCable Dynamic Quality of Service [35], then the parameters in NCS [36] would be used instead of the reservation direction, confirmation and reservation sharing parameters described here.

#### 2.11.2. Parameter Encoding

The Local Connection Options for the "RES" package consist of the following:

- \* The resource reservation parameter, encoded as the keyword "r", followed by a colon and the value "g" (guaranteed service), "cl" (controlled load) or "be" (best effort).
- \* The reservation direction parameter, encoded as the keyword "r-dir" followed by a colon and the value "sendonly", "recvonly" or "sendrecv".
- \* The reservation confirmation parameter, encoded as the keyword "r-cnf" followed by a colon and the value "none", "sendonly", "recvonly" or "sendrecv".
- \* The resource sharing parameter, encoded as the keyword "r-sh" followed by a colon and either:
  - \* The wild-card character "\$" indicating this connection, indicating future plans to share resources with this connection, or
  - \* A connection ID, indicating a request to share resources with the connection having the specified connection ID (and all other connections sharing resources with that connection), or

- \* An empty value (i.e., "r-sh:" with no value indicated), indicating a request to no longer share the resources of this connection with other connections

Note that normally local connection options that are associated with a package have the package prefix included as per the package extension rules in [1]. The local connection options in the "RES" package are exceptions. The package prefix is not included in the case of the "RES" package because it was created before the extension rules in [1] were defined.

### 2.11.3. Events

The following events are included as part of the resource reservation package:

Symbol	Definition	R	S	Duration
re	Resource Error	C		
rl	Resource Lost	C		

#### Resource Error (re):

This is an indication that an error in the resource reservation occurred during the life of the connection. This event is not requested with a parameter, but is reported with a parameter (see possible values below). This event may or may not indicate the permanent loss of the reservation (i.e., any error associated with the reservation whether permanent or temporary will be reported). If requested on an endpoint (without specifying the connection ID), the request refers to all present and future connections on that endpoint. When reported, the connectionID is always supplied along with a reason for the error indicated as a parameter. One of the following possible reasons for loss MUST be included as the parameter when the event is reported:

- "resvterr" is used to indicate that a ResvErr message was received.
- "patherr" is used to indicate that a PathErr message was received.
- "other"

In addition to a parameter indicating one of the reasons above, additional information on the type of error MAY be included as a second parameter in the form of a quoted string.

Example report might include:

```
O: res/rl@0A3F58(resvrr)
```

or

```
O: res/rl@0A3F58(resvrr, "some additional commentary")
```

Note that this event will not be reported if an error occurs while a resource reservation is initially being set up (i.e., the event was only reported as a result of an error that occurred after the reservation was set up).

#### Resource Lost (rl):

Loss of reservation during the life of a connection can be reported by using the "rl" event. This event is not requested with a parameter, but is reported with a parameter (see below for possible values). If requested on an endpoint (without specifying the connection ID), the request refers to all present and future connections on that endpoint.

When reported, the connectionID is always supplied along with a reason for the loss indicated as a parameter. One of the following possible reasons for loss MUST be supplied as the parameter when the event is reported:

- "resvtear" indicating that the reservation loss was indicated by ResvTear message.
- "pathtear" indicating that the reservation loss was indicated by PathTear message.
- "other"

In addition to a parameter indicating one of the reasons above, additional information on the type of error MAY be included as a second parameter in the form of a quoted string.

Example report might include:

```
O: res/rl@0A3F58(ResvTear)
```

or

```
O: res/rl@0A3F58(ResvTear, "some other commentary")
```

Note that this event will not be reported if an error occurs while a resource reservation is initially being set up (i.e., the event is only reported if the reservation was lost after it was initially set up).

## 2.12. Announcement Server Package

Package Name: A

Version: 1

Symbol	Definition	R	S	Duration
ann(url)	Play an Announcement			TO, C variable
oc	Operation Complete	x		
of	Operation Failure	x		

Changes from the previous version: change to conform to standard reporting of operation failure and operation complete events.

The announcement signal is qualified by a URL name:

S: ann(http://scripts.example.net/all-lines-busy.au)

The URL name MAY be followed by a list of initial parameters, separated by commas. However, standard parameters are not included as part of this package definition (Note: use of additional parameters is optional and would result in a proprietary interface).

The gateway SHOULD support one or more standard URL schemes such as:

- \* file, http, ftp (RFC 1738 [28]), which indicate where the audio file is located (where to load the file from before playing the audio file on the gateway).
- \* RTSP URL (section 3.2 of RFC 2326 [29]), which in this case allows the media gateway to directly initiate playing of the announcement via an RTSP server.

The pre-condition for a successful response (return code of "200") is correct syntax and capability (support is available for this request). Standard MGCP return codes apply in the case of failure. Further indications of failure are provided in the operation failure event as a comment after the name of the failed event in the form of a quoted string.

If the announcement cannot be played out for a reason determined after a successful response to the request has been provided, an operation failure event will be returned. The failure MAY be explained by some commentary (in the form of a quoted string), as in:

```
O: a/of(a/ann,"file not found")
```

The "operation complete" event will be detected when the announcement is played out.

```
O: a/oc(a/ann)
```

### 2.13. Script Package

Package Name: Script

Version: 1

Symbol	Definition	R	S	Duration
ir(..)	Intermediate Results/Req.	x	BR	
java(url,...)	Load & Run java script		TO	variable
oc	operation complete	x		
of	operation failure	x		
perl(url,...)	Load & Run perl script		TO	variable
tcl(url,...)	Load & Run TCL script		TO	variable
vxml(url,...)	Load & Run VXML doc.		TO	variable
xml(url,...)	Load & Run XML script		TO	variable

Changes from the previous version of the package: "vxml" was added as a language type for loading and running VXML documents; change to conform with standard reporting of operation failure and operation complete events; addition of "ir" event.

The current definition defines keywords for the most common languages. More languages may be defined in later versions of this package.

The "signal" specifying the scripting language is parameterized with a URL indicating the location of the script. The URL parameter MAY be optionally followed by a comma-separated list of arguments as initial parameters to use in running the script. URL schemes may include file ftp, or http schemes with syntax according to RFC 2396 [30]. As an example:

```
S: script/vxml(ftp://ftp.example.net/credit-card.vxml,arg1,arg2,
               ...,argn)
```

The argument list "arg1,arg2,...,argn" is passed to the script/document as a list of initial parameters.

The pre-condition for a successful response (return code of "200") is correct syntax and capability (support is available for this request). Standard MGCP return codes apply in the case of failure. Some further (non-application/script specific) failure indications MAY be provided in the operation failure event as a comment in the form of a quoted string following the name of the failed event.

#### Example

```
O: script/of(script/vxml,"file not found")
```

The script produces an output, which consists of one or several text strings, separated by commas. This provides the return-status of the script as well as return parameters (if there are any)

```
O: script/oc(script/vxml,return-status=<status>,
             name1=value1,name2=value2,...)
```

where <status> can have one of the values "success" or "failure". This is then followed by output parameters as a comma-separated list of name-value pairs.

Intermediate Result/Request (ir(<params>)):

This provides a way for:

- \* The script to inform the Call Agent of intermediate results (e.g., a case where it is important because of timing concerns to inform the Call Agent prior to operation complete).
- \* The script to request some information from the Call Agent.
- \* The Call Agent to inform the script of some event or information that may be important for the operation of the script (in this case "ir" is used as a signal).

Parameters (i.e., <params>) SHOULD be a comma-separated list of name-value pairs e.g., ir(name1=value1,name2=value2,...). The Call Agent MAY include event parameters when it requests this event, in which case, the MGCP syntax requirements require that the action be specified (e.g., "R: ir(N)(name1=value1,name2=value2,...)").

If the Call Agent requests "ir" as a signal, at least one parameter MUST be provided.

When requesting the "ir" signal, the Call Agent MUST also repeat the original script signal. This is in order to be consistent with the semantics of TO signals in MGCP (i.e., if the original "script" signal is not included, then the signal/script will be stopped). The only problem with this is that there is a possible race condition in which a request to send an "ir" signal could occur just as the script stopped. In order to avoid this confusion, the following is RECOMMENDED: when the script signal is included with an "ir" signal, include a parameter (of the script signal) to indicate that this is not a new instance of the script i.e., if there is no script executing at the present time do not start executing a new one.

The "ir" signal is only associated with an executing script. If none are running when a request for the event/signal is made, or if a new script request is not included with the request, then the "ir" signal/event will not be executed (i.e., the "ir" event with its parameters is passed to an existing script for parsing and execution and is considered opaque as far as MGCP as concerned. If no such script exists, response code "800" will be returned, indicating that the script is not executing).

The following response code is associated with this package:

Code	Text	Explanation
800	Script not Executing	Request for "ir" signal or event but no script is executing at the time the request was received.

Note that package specific error codes include the package name following the error code. For example, if error code 800 occurs in response to a request with a transaction ID of 1001, it would be sent as:

800 1001 /SCRIPT

### 3. IANA Considerations

The following packages and their versions have been registered with IANA as per the instructions in [1].

Package Title	Name	Version
-----	----	-----
Announcement	A	1
DTMF	D	1
Digit Map Extension	DM1	0
Media Format	FM	0
Generic	G	1
Handset	H	1
Line	L	1
RTP	R	1
Resource Reservation	RES	0
Script	SCRIPT	1
Supplementary Tones	SST	0
Signal List	SL	0
Trunk	T	1

The following extension digit map letter has been registered with IANA:

Package Letter
-----
DM1      P

The following Local Connections have been registered with IANA:

Field	Name
-----	-----
Media Format	fntp
Reservation Confirmation	r-cnf
Reservation Direction	r-dir
Resource Sharing	r-sh

### 4. Security Considerations

The MGCP packages contained in this document do not require any further security considerations beyond those indicated in the base MGCP specification [1].

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## 7. Authors' Addresses

Bill Foster  
Cisco Systems

Phone: +1 250 758 9418  
EMail: [bfoster@cisco.com](mailto:bfoster@cisco.com)

Flemming Andreasen  
Cisco Systems  
Edison, NJ 08837

EMail: [fandreas@cisco.com](mailto:fandreas@cisco.com)

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