

SIP—INFO Method for DTMF Tone Generation

The SIP—INFO Method for DTMF Tone Generation feature uses the Session Initiation Protocol (SIP) INFO method to generate dual tone multifrequency (DTMF) tones on the telephony call leg. SIP info methods, or request message types, request a specific action be taken by another user agent (UA) or proxy server. The SIP INFO message is sent along the signaling path of the call. Upon receipt of a SIP INFO message with DTMF relay content, the gateway generates the specified DTMF tone on the telephony end of the call.

Prerequisites for SIP—INFO Method for DTMF Tone Generation

Cisco Unified Border Element

- Cisco IOS Release 12.2(11)T or a later release must be installed and running on your Cisco Unified Border Element.

Cisco Unified Border Element (Enterprise)

- Cisco IOS XE Release 2.5 or a later release must be installed and running on your Cisco ASR 1000 Series Router.

Information About SIP—INFO Method for DTMF Tone Generation

The SIP—INFO Method for DTMF Tone Generation feature is always enabled, and is invoked when a SIP INFO message is received with DTMF relay content. This feature is related to the [DTMF Events Through SIP Signaling](#) feature, which allows an application to be notified about DTMF events using SIP NOTIFY messages. Together, the two features provide a mechanism to both send and receive DTMF digits along the signaling path. For more information on sending DTMF event notification using SIP NOTIFY messages, refer to the [DTMF Events Through SIP Signaling](#) feature.

How to Review SIP INFO Messages

The SIP INFO method is used by a UA to send call signaling information to another UA with which it has an established media session. The following example shows a SIP INFO message with DTMF content:

```
INFO sip:2143302100@172.17.2.33 SIP/2.0
Via: SIP/2.0/UDP 172.80.2.100:5060
From: <sip:9724401003@172.80.2.100>;tag=43
To: <sip:2143302100@172.17.2.33>;tag=9753.0207
Call-ID: 984072_15401962@172.80.2.100
CSeq: 25634 INFO
Supported: 100rel
Supported: timer
Content-Length: 26
Content-Type: application/dtmf-relay

Signal= 1
Duration= 160
```

This sample message shows a SIP INFO message received by the gateway with specifics about the DTMF tone to be generated. The combination of the “From”, “To”, and “Call-ID” headers identifies the call leg. The signal and duration headers specify the digit, in this case 1, and duration, 160 milliseconds in the example, for DTMF tone play.

Prerequisites

The following are general prerequisites for SIP functionality:

- Ensure that the gateway has voice functionality that is configured for SIP.
- Establish a working IP network.
- Configure VoIP.

Restrictions

The SIP—INFO Method for DTMF Tone Generation feature includes the following signal duration parameters:

- Minimum signal duration is 100 milliseconds (ms). If a request is received with a duration less than 100 ms, the minimum duration of 100 ms is used by default.
- Maximum signal duration is 5000 ms. If a request is received with a duration longer than 5000 ms, the maximum duration of 5000 ms is used by default.
- If no duration parameter is included in a request, the gateway defaults to a signal duration of 250 ms.

Configuring for SIP—INFO Method for DTMF Tone Generation

You cannot configure, enable, or disable this feature. No configuration tasks are required to configure the SIP - INFO Method for DTMF Tone Generation feature. The feature is enabled by default.

Troubleshooting Tips

You can display SIP statistics, including SIP INFO method statistics, by using the **show sip-ua statistics** and **show sip-ua status** commands in privileged EXEC mode. See the following fields for SIP INFO method statistics:

- OkInfo 0/0, under SIP Response Statistics, Success, displays the number of successful responses to an INFO request.
- Info 0/0, under SIP Total Traffic Statistics, displays the number of INFO messages received and sent by the gateway.

The following is sample output from the **show sip-ua statistics** command:

```
Router# show sip-ua statistics

SIP Response Statistics (Inbound/Outbound)
Informational:
  Trying 1/1, Ringing 0/0,
  Forwarded 0/0, Queued 0/0,
  SessionProgress 0/1
Success:
  OkInvite 0/1, OkBye 1/0,
  OkCancel 0/0, OkOptions 0/0,
  OkPrack 0/0, OkPreconditionMet 0/0
  OkSubscribe 0/0, OkNotify 0/0,
  OkInfo 0/0, 202Accepted 0/0
Redirection (Inbound only):
  MultipleChoice 0, MovedPermanently 0,
  MovedTemporarily 0, SeeOther 0,
  UseProxy 0, AlternateService 0
Client Error:
```

```

        BadRequest 0/0, Unauthorized 0/0,
        PaymentRequired 0/0, Forbidden 0/0,
        NotFound 0/0, MethodNotAllowed 0/0,
        NotAcceptable 0/0, ProxyAuthReqd 0/0,
        ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
        LengthRequired 0/0, ReqEntityTooLarge 0/0,
        ReqURITooLarge 0/0, UnsupportedMediaType 0/0,
        BadExtension 0/0, TempNotAvailable 0/0,
        CallLegNonExistent 0/0, LoopDetected 0/0,
        TooManyHops 0/0, AddrIncomplete 0/0,
        Ambiguous 0/0, BusyHere 0/0,
        BadEvent 0/0
    Server Error:
        InternalError 0/0, NotImplemented 0/0,
        BadGateway 0/0, ServiceUnavail 0/0,
        GatewayTimeout 0/0, BadSipVer 0/0
    Global Failure:
        BusyEverywhere 0/0, Decline 0/0,
        NotExistAnywhere 0/0, NotAcceptable 0/0
SIP Total Traffic Statistics (Inbound/Outbound)
    Invite 0/0, Ack 0/0, Bye 0/0,
    Cancel 0/0, Options 0/0,
    Prack 0/0, Comet 0/0,
    Subscribe 0/0, Notify 0/0,
    Refer 0/0, Info 0/0
Retry Statistics
    Invite 0, Bye 0, Cancel 0, Response 0, Notify 0

```

The following is sample output from the **show sip-ua status** command:

```

Router# show sip-ua status

SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED
SIP User Agent bind status(signaling): DISABLED
SIP User Agent bind status(media): DISABLED
SIP max-forwards : 6
SIP DNS SRV version: 2 (rfc 2782)
SDP application configuration:
    Version line (v=) required
    Owner line (o=) required
    Session name line (s=) required
    Timespec line (t=) required
Media supported: audio image
Network types supported: IN
Address types supported: IP4
Transport types supported: RTP/AVP udptl

```