DTMF Events through SIP Signaling

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The DTMF Events through SIP Signaling feature provides the following:

- DTMF event notification for SIP messages.
- Capability of receiving hookflash event notification through the SIP NOTIFY method.
- Third-party call control, or other signaling mechanisms, to provide enhanced services, such as calling card and messaging services.
- Communication with the application outside of the media connection.

The DTMF Events through SIP Signaling feature allows telephone event notifications to be sent through SIP NOTIFY messages, using the SIP SUBSCRIBE/NOTIFY method as defined in the Internet Engineering Task Force (IETF) draft, SIP-Specific Event Notification.

The feature also supports sending DTMF notifications based on the IETF draft: Signaled Telephony Events in the Session Initiation Protocol (SIP) (draft-mahy-sip-signaled-digits-01.txt).

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the Feature Information Table at the end of this document.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.

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Prerequisites for DTMF Events through SIP Signaling

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.

Restrictions for DTMF Events through SIP Signaling

The DTMF Events through SIP Signaling feature adds support for sending telephone-event notifications via SIP NOTIFY messages from a SIP gateway. The events for which notifications are sent out are DTMF events from the local Plain Old Telephone Service (POTS) interface on the gateway. Notifications are not sent for DTMF events received in the Real-Time Transport Protocol (RTP) stream from the recipient user agent.

Configuring DTMF Events through SIP Signaling

To configure the DTMF Events through SIP Signaling feature, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **timers notify** *number*
5. **retry notify** *number*
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enters privileged EXEC mode or any other security level set by a system administrator. • Enter your password if prompted.

	Command or Action	Purpose
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters SIP user-agent configuration mode.
Step 4	**timers notify** *number* **Example:** `Router(config-sip-ua)# timers notify 100`	Sets the amount of time that the user agent waits before retransmitting the Notify message. The argument is as follows: • *number* --Time, in milliseconds, to wait before retransmitting. Range: 100 to 1000. Default: 500.
Step 5	**retry notify** *number* **Example:** `Router(config-sip-ua)# retry notify 6`	Sets the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or Refer request. The argument is as follows: • *number* --Number of retries. Range: 1 to 10. Default: 10.
Step 6	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits the current mode.

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Verifying SIP DTMF Support

To verify SIP DTMF support, perform the following steps as appropriate (commands are listed in alphabetical order).

SUMMARY STEPS

1. **show running-config**
2. **show sip-ua retry**
3. **show sip-ua statistics**
4. **show sip-ua status**
5. **show sip-ua timers**
6. **show voip rtp connections**
7. **show sip-ua calls**

DETAILED STEPS

Step 1 **show running-config**

Use this command to show dial-peer configurations.

The following sample output shows that the **dtmf-relay sip-notify** command is configured in dial peer 123:

Example:

```
Router# show running-config
.
.
.
dial-peer voice 123 voip
 destination-pattern [12]...
 monitor probe icmp-ping
 session protocol sipv2
 session target ipv4:10.8.17.42
 dtmf-relay sip-notify
```

The following sample output shows that DTMF relay and NTE are configured on the dial peer.

Example:

```
Router# show running-config
!
dial-peer voice 1000 pots
 destination-pattern 4961234
 port 1/0/0
!
dial-peer voice 2000 voip
 application session
 destination-pattern 4965678
 session protocol sipv2
 session target ipv4:192.0.2.34
 dtmf-relay rtp-nte
! RTP payload type value = 101 (default)
!
dial-peer voice 3000 voip
 application session
 destination-pattern 2021010101
 session protocol sipv2
 session target ipv4:192.0.2.34
 dtmf-relay rtp-nte
 rtp payload-type nte 110
! RTP payload type value = 110 (user assigned)
!
```

Step 2 **show sip-ua retry**

Use this command to display SIP retry statistics.

Example:

```
Router# show sip-ua retry
SIP UA Retry Values
invite retry count = 6 response retry count = 1
bye retry count = 1 cancel retry count = 1
prack retry count = 10 comet retry count = 10
reliable 1xx count = 6 notify retry count = 10
```

Step 3 **show sip-ua statistics**

Use this command to display response, traffic, and retry SIP statistics.

Tip To reset counters for the **show sip-ua statistics**display, use the **clear sip-ua statistics** command.

Example:

```
Router# show sip-ua statistics
SIP Response Statistics (Inbound/Outbound)
Informational:
Trying 4/2, Ringing 2/1,
Forwarded 0/0, Queued 0/0,
SessionProgress 0/0
Success:
OkInvite 1/2, OkBye 0/1,
OkCancel 1/0, OkOptions 0/0,
OkPrack 2/0, OkPreconditionMet 0/0,
OkNotify 1/0, 202Accepted 0/1
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
RequestCancel 1/0, NotAcceptableMedia 0/0
Server Error:
InternalError 0/1, NotImplemented 0/0,
BadGateway 0/0, ServiceUnavail 0/0,
GatewayTimeout 0/0, BadSipVer 0/0,
PreCondFailure 0/0
Global Failure:
BusyEverywhere 0/0, Decline 0/0,
NotExistAnywhere 0/0, NotAcceptable 0/0
SIP Total Traffic Statistics (Inbound/Outbound) /* Traffic Statistics
Invite 3/2, Ack 3/2, Bye 1/0,
Cancel 0/1, Options 0/0,
Prack 0/2, Comet 0/0,
Notify 0/1, Refer 1/0
Retry Statistics                             /* Retry Statistics
Invite 0, Bye 0, Cancel 0, Response 0,
Prack 0, Comet 0, Reliable1xx 0, Notify 0
```

Following is sample output verifying configuration of the SIP INFO Method for DTMF Tone Generation feature:

Example:

```
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
OkSubscibe 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
```

Step 4 **show sip-ua status**

Use this command to display status for the SIP user agent.

Example:

```
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
```

The following sample output shows that the time interval between consecutive NOTIFY messages for a telephone event is the default of 2000 ms:

Example:

```
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 early-media for 180 responses with SDP: ENABLED
SIP max-forwards : 6
SIP DNS SRV version: 2 (rfc 2782)
NAT Settings for the SIP-UA
Role in SDP: NONE
Check media source packets: DISABLED
Maximum duration for a telephone-event in NOTIFYs: 2000 ms
SIP support for ISDN SUSPEND/RESUME: ENABLED
Redirection (3xx) message handling: ENABLED
 SDP application configuration:
 Version line (v=) required
 Owner line (o=) required
 Timespec line (t=) required
 Media supported: audio image
 Network types supported: IN
 Address types supported: IP4
 Transport types supported: RTP/AVP udptl
```

The following sample output shows configuration of the SIP INFO Method for DTMF Tone Generation feature:

Example:

```
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
```

Step 5 **show sip-ua timers**

Use this command to display the current settings for SIP user-agent timers.

Example:

```
Router# show sip-ua timers
SIP UA Timer Values (millisecs)
trying 500, expires 300000, connect 500, disconnect 500
comet 500, prack 500, rel1xx 500, notify 500
```

Step 6 **show voip rtp connections**

Use this command to show local and remote Calling ID and IP address and port information.

Step 7 **show sip-ua calls**

Use this command to ensure the DTMF method is SIP-KPML.

The following sample output shows that the DTMF method isSIP-KPML.

Example:

```
router# show sip-ua calls
SIP UAC CALL INFO
Call 1
SIP Call ID                : 57633F68-2BE011D6-8013D46B-B4F9B5F6@172.18.193.251
   State of the call       : STATE_ACTIVE (7)
   Substate of the call    : SUBSTATE_NONE (0)
   Calling Number          :
   Called Number           : 8888
   Bit Flags               : 0xD44018 0x100 0x0
   CC Call ID              : 6
   Source IP Address (Sig ): 192.0.2.1
   Destn SIP Req Addr:Port : 192.0.2.2:5060
   Destn SIP Resp Addr:Port: 192.0.2.3:5060
   Destination Name        : 192.0.2.4.250
   Number of Media Streams : 1
   Number of Active Streams: 1
   RTP Fork Object         : 0x0
   Media Mode              : flow-through
   Media Stream 1
     State of the stream      : STREAM_ACTIVE
     Stream Call ID           : 6
     Stream Type              : voice-only (0)
     Negotiated Codec         : g711ulaw (160 bytes)
    Codec Payload Type       : 0
     Negotiated Dtmf-relay    : sip-kpml
     Dtmf-relay Payload Type  : 0
     Media Source IP Addr:Port: 192.0.2.5:17576
     Media Dest IP Addr:Port  : 192.0.2.6:17468
     Orig Media Dest IP Addr:Port : 0.0.0.0:0
   Number of SIP User Agent Client(UAC) calls: 1
SIP UAS CALL INFO
   Number of SIP User Agent Server(UAS) calls: 0
```

Troubleshooting Tips

- To enable debugging for RTP named-event packets, use the **debug voip rtp** command.
- To enable KPML debugs, use the **debug kpml** command.
- To enable SIP debugs, use the **debug ccsip** command.
- Collect debugs while the call is being established and during digit presses.
- If an established call is not sending digits through KPML, use the **show sip-ua calls** command to ensure SIP-KPML is included in the negotiation process.

Feature Information for DTMF Events through SIP Signaling

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

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ISR Feature History Entry.

Table 1 Feature Information for Configuring DTMF Events through SIP Signaling

Feature Name	Releases	Feature Information
DTMF Events through SIP Signaling	12.2(11)T 12.2(8)YN 12.2(15)T 12.2(11)YV 12.2(11)T,	The DTMF Events through SIP Signaling feature provides the following: • DTMF event notification for SIP messages. • Capability of receiving hookflash event notification through the SIP NOTIFY method. • Third-party call control, or other signaling mechanisms, to provide enhanced services, such as calling card and messaging services. • Communication with the application outside of the media connection. The following commands were introduced or modified: **timers notify** and **retry notify**.

ASR Feature History Entry.

Table 2 Feature Information for Configuring DTMF Events through SIP Signaling

Feature Name	Releases	Feature Information
DTMF Events through SIP Signaling	Cisco IOS XE Release 2.5	The DTMF Events through SIP Signaling feature provides the following: • DTMF event notification for SIP messages. • Capability of receiving hookflash event notification through the SIP NOTIFY method. • Third-party call control, or other signaling mechanisms, to provide enhanced services, such as calling card and messaging services. • Communication with the application outside of the media connection. The following commands were introduced or modified: **timers notify** and **retry notify**.