DTMF Events through SIP Signaling

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).

The following pages provide information on the DTMF Events through SIP Signaling feature:

• Finding Feature Information, page 1
• Prerequisites for DTMF Events through SIP Signaling, page 2
• Restrictions for DTMF Events through SIP Signaling, page 2
• DTMF Dialing, page 2
• NOTIFY Messages, page 2
• Configuring DTMF Events through SIP Signaling, page 3
• Troubleshooting Tips, page 9
• Feature Information for DTMF Events through SIP Signaling, page 9

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and 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.
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.

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.

DTMF Dialing

DTMF dialing consists of simultaneous voice-band tones generated when a button is pressed on a telephone. The use of DTMF signaling for this feature enables support for advanced telephony services. Currently there are a number of application servers and service creation platforms that do not support media connections. To provide value-added services to the network, these servers and platforms need to be aware of signaling events from a specific participant in the call. Once the server or platform is aware of the DTMF events that are being signaled, it can use third-party call control, or other signaling mechanisms, to provide enhanced services. Examples of the types of services and platforms that are supported by this feature are various voice web browser services, Centrex switches or business service platforms, calling card services, and unified message servers. All of these applications require a method for the user to communicate with the application outside of the media connection. The DTMF Events Through SIP Signaling feature provides this signaling capability. This feature is related to the SIP INFO Method for DTMF Tone Generation feature, which adds support for out-of-band DTMF tone generation using the SIP INFO method. Together the two features provide a mechanism to both send and receive DTMF digits along the signaling path.

NOTIFY Messages

The SIP event notification mechanism uses NOTIFY messages to signal when certain telephony events take place. In order to send DTMF signals through NOTIFY messages, the gateway notifies the subscriber when DTMF digits are signaled by the originator. The notification contains a message body with a SIP response status line.
The following sample message shows a NOTIFY message from the Notifier letting the Subscriber know that the subscription is completed. The combination of the From, To, and Call-ID headers identifies the call leg. The Events header specifies the event type being signaled, and the Content-Type specifies the Internet media type. The Content-Length header indicates the number of octets in the message body.

```
NOTIFY sip:subscriber@example1.com SIP/2.0
Via: SIP/2.0/UDP example2.com:5060
From: Notifier <sip:notifier@example2.com>;tag=5678-EFGH
To: Subscriber <sip:subscriber@example1.com>;tag=1234-ABCD
Call-ID: 12345@example2.com
CSeq: 104 NOTIFY
Contact: Notifier <sip:notifier@example2.com>
Events: telephone-event;rate=1000
Content-Type: audio/telephone-event
Content-Length: 4
```

## 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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enters privileged EXEC mode or any other security level set by a system administrator.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Device&gt; enable</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Device# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> sip-ua</td>
<td>Enters SIP user-agent configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Device(config)# sip-ua</td>
<td></td>
</tr>
</tbody>
</table>
### Purpose

**Command or Action**

<table>
<thead>
<tr>
<th>Step 4</th>
<th>timers notify number</th>
</tr>
</thead>
</table>

**Example:**

Device(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.

<table>
<thead>
<tr>
<th>Step 5</th>
<th>retry notify number</th>
</tr>
</thead>
</table>

**Example:**

Device(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:


<table>
<thead>
<tr>
<th>Step 6</th>
<th>exit</th>
</tr>
</thead>
</table>

**Example:**

Device(config-sip-ua)# exit

Exits the current mode.

### 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:

Device# 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:

Device# 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-nce
    ! 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-nce
    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:

Device# 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:

Device# show sip-ua statistics
SIP Response Statistics (Inbound/Outbound)
Following is sample output verifying configuration of the SIP INFO Method for DTMF Tone Generation feature:

```
Example:
Device# show sip-ua statistics
```

Cisco Unified Border Element (Enterprise) Protocol-Independent Features and Setup Configuration Guide,
Cisco IOS XE Release 3S
Step 4  
**show sip-ua status**

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

**Example:**

```
Device# 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:**

```
Device# 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:

Device# 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:

Device# show sip-ua timers
SIP UA Timer Values (millsecs)
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 is SIP-KPML.

Example:

Device# show sip-ua calls
SIP UAC CALL INFO
Call 1
SIP Call ID : 57633F68-2BE0311D6-8013D46B-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
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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Table 1: Feature Information for Configuring DTMF Events through SIP Signaling

<table>
<thead>
<tr>
<th>Feature Name</th>
<th>Releases</th>
<th>Feature Information</th>
</tr>
</thead>
</table>
| 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:  
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- Communication with the application outside of the media connection.  

The following commands were introduced or modified: timers notify and retry notify. |
| DTMF Events through SIP Signaling | Cisco IOS XE Release 2.5 | The DTMF Events through SIP Signaling feature provides the following:  
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