Appendix A: Preparing Cisco Unified SRST Support for SIP

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Cisco Unified Survivable Remote Site Telephony (SRST) supports incoming and outgoing Session Initiation Protocol (SIP) calls to and from Cisco Unified IP phones and router voice gateway voice ports, but does not support direct attachment of SIP phones to Cisco Unified SRST. SIP may be used in situations where the Cisco Unified SRST Router is separate from the PSTN gateway and the SRST and PSTN gateways are linked together using SIP (instead of H.323).

Special configurations to support SIP calls are described in this appendix. For more information about SIP, see the *Cisco IOS SIP Configuration Guide*.

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DTMF Relay for SIP Applications and Voice Mail

DTMF relay for SIP applications can be used in two voice-mail situations:

DTMF Relay Using SIP RFC 2833

Cisco Unified Skinny Client Control Protocol (SCCP) Phones, such as those used with Cisco Unified SRST systems, provide only out-of-band DTMF digit indications. To enable SCCP phones to send digit information to remote SIP-based IVR and voice-mail applications, Cisco Unified SRST 3.2 and later versions provide conversion from the out-of-band SCCP digit indication to the SIP standard for DTMF relay, which is RFC 2833. You select this method in the SIP VoIP dial peer using the **dtmf-relay rtp-nte** command.

The SIP DTMF relay method is needed in the following situations:

- When SIP is used to connect a Cisco Unified SRST system to a remote SIP-based IVR or voice-mail application, such as Cisco Unity.
- When SIP is used to connect a Cisco Unified SRST system to a remote SIP-PSTN voice gateway that goes through the PSTN to a voice-mail or IVR application.

Note The need to use out-of-band DTMF relay conversion is limited to SCCP phones. SIP phones natively support in-band DTMF relay as specified in RFC 2833.

To enable SIP DTMF relay using RFC 2833, the commands in this section must be used on both originating and terminating gateways.

SUMMARY STEPS

1. **dial-peer voice** *tag* **voip**
2. **dtmf-relay rtp-nte**
3. **exit**
4. **sip-ua**
5. **notify telephone-event max-duration** *time*
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 2 voip`	Enters dial-peer configuration mode.
Step 2	**dtmf-relay rtp-nte** **Example:** `Router(config-dial-peer)# dtmf-relay rtp-nte`	Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with the Named Telephone Event (NTE) payload type.
Step 3	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial-peer configuration mode.
Step 4	**sip-ua** **Example:** `Router(config)# sip-ua`	Enables SIP user-agent configuration mode.

	Command or Action	Purpose
Step 5	**notify telephone-event max-duration** *time* **Example:** `Router(config-sip-ua)# notify telephone-event max-duration 2000`	Configures the maximum time interval allowed between two consecutive NOTIFY messages for a single DTMF event. • **max-duration** *time*: Time interval between consecutive NOTIFY messages for a single DTMF event, in milliseconds. Range is from 500 to 3000. Default is 2000.
Step 6	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits SIP user-agent configuration mode.

Troubleshooting Tips

The dial-peer section of the **show running-config** command output displays DTMF relay status when it is configured, as shown in this excerpt:

```
dial-peer voice 123 voip
 destination-pattern [12]...
 monitor probe icmp-ping
 session protocol sipv2
 session target ipv4:10.8.17.42
 dtmf-relay rtp-nte
```

DTMF Relay Using SIP Notify (Nonstandard)

To use voice mail on a SIP network that connects to a Cisco Unity Express system, use a nonstandard SIP Notify format. To configure the Notify format, use the **sip-notify** keyword with the **dtmf-relay** command. Using the **sip-notify** keyword may be required for backward compatibility with Cisco SRST Versions 3.0 and 3.1.

SUMMARY STEPS

1. **dial-peer voice** *tag* **voip**
2. **dtmf-relay sip-notify**
3. **exit**
4. **sip-ua**
5. **notify telephone-event max-duration** *time*
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 2 voip`	Enters dial-peer configuration mode.
Step 2	**dtmf-relay sip-notify** **Example:** `Router(config-dial-peer)# dtmf-relay sip-notify`	Forwards DTMF tones using SIP NOTIFY messages.
Step 3	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial-peer configuration mode.
Step 4	**sip-ua** **Example:** `Router(config)# sip-ua`	Enables SIP user-agent configuration mode.
Step 5	**notify telephone-event max-duration** *time* **Example:** `Router(config-sip-ua)# notify telephone-event max-duration 2000`	Configures the maximum time interval allowed between two consecutive NOTIFY messages for a single DTMF event. • **max-duration** *time*: Time interval between consecutive NOTIFY messages for a single DTMF event, in milliseconds. Range is from 500 to 3000. Default is 2000.
Step 6	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits SIP user-agent configuration mode.

Troubleshooting Tips

The **show sip-ua status** command output displays the time interval between consecutive NOTIFY messages for a telephone event. In the following example, the time interval is 2000 ms.

```
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
```

Where to Go Next

For information about SRST and SIP, see the *Cisco Unified SIP SRST System Administrator Guide*.

For information about monitoring and maintaining Cisco Unified SRST, go to the "Monitoring and Maintaining Cisco Unified SRST" section on page 223.

For additional information, see the "Additional References" section on page 44 in the "Overview of Cisco Unified SRST" section on page 31.