| | | |
|---|---|---|
| Version 9.0 | 15.2(2)T | • |

## DETAILED STEPS

# transfer-pattern blocked

## conference transfer-pattern

# New Features in Cisco Unified SRST Version 4.2(1)

Cisco Unified SRST Version 4.2(1) introduces the following new features:

- Enhancements for

# E1 R2 Signaling Support

# New Features in Cisco SRST Version 2.02

CHAPTER 1

# Cisco Unified SRST Feature Overview

This chapter describes Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) and what it does. It also

**Note** During a WAN connection failure, when Cisco Unified SRST

# 1

# Cisco Unified SRST Licenses

You should purchase a Cisco Unified SRST license th

- Cisco Unified SRST Permanent License or
- 

## SUMMARY STEPS

5. **enable**
6. **configure terminal**
7.

The following example provides interface configuration for IPv6 supported on Unified SRST:

```
configure terminal
TddTJss 2fig:420:54FF:13::w [(T312:82/1196 TD [(inte481igu)7. ipon ew [(Tnable 7.98 0 0 9.96 36 704.34 Tm 0 g617205 Tc .
```

# Related Documents

| Related Topic | Documents |
|---|---|
| Cisco IOS voice product configuration | • *Cisco IOS Voice Configuration Library* |

## Standards

## MIBs

## RFCs

## Technical Assistance

# Obtaining Documentation, Obtaining Support, and Security Guidelines

For information on obtaining documentation, obtaining support, providing documentation feedback, security guidelines, and also recommended aliases and general Cisco documents, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at

http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html.

*Table 2-2 ESRST Scales Supported on All Platforms*

## Example: ESRST Scale Increase

# Information About Setting Up the Network

When the WAN link fails, the Cisco Unified IP Phones detect that they are no

### SUMMARY STEPS

1.

### DETAILED STEPS

## Specifying Keepalive Intervals

The keepalive interval is the period of time between keepalive messages sent by a network device. A keepalive message is a message sent by one network device to inform another network device that the virtual circuit between the two is still active.

### DETAILED STEPS

## Examples

CHAPTER 4

CHAPTER

# Troubleshooting

To troubleshoot your Cisco Unified SRST configuration, use the following commands:

- To set keepalive debugging for Cisco IP phones, use the **debug ephone keepalive** command.
- To set registration debugging for Cisco IP phones, use the **debug ephone register** command.
- To set state debugging for Cisco IP phones, use the **debug ephone state** command.
-

## Configuring Backup Registrar Service to SIP Phones (Using Optional Commands)

## DETAILED STEPS

| Command or Action | Purpose |
|---|---|

**Step 6** **`show dial-peer voice`**

**Example:**

```
Router# show dial-peer voice
VoiceOverIpPeer40036
peer type = voice, information type = voice,
description = `',
tag = 40036, destination-pattern = `91011',
answer-address = `', preference=1,
CLID Restriction = None
CLID Network Number = `'
CLID Second Number sent
source carrier-id = `', target carrier-id = `',
source trunk-group-label = `', target
trunk-group-label = `',
numbering Type = `unknown'
group = 40036, Admin state is up, Operation state is
up,
incoming called-number = `', connections/maximum =
0/unlimited,
! Default output for incoming called-number command
DTMF Relay = disabled,
modem transport = system,
huntstop = disabled,
in bound application associated: 'DEFAULT'
```

# Prerequisites for Configuring SIP SRST Features Using Back-to-Back User Agent Mode

- Complete the prerequisites documented in the "Prerequisites for Configuring Cisco Unified SIP SRST" section on page 9 section in the "" section on page 1.
-

*Table 7-1 Version 3.4 New or Enhanced Commands for Cisco Unified SRST and Cisco Unified CME (Sorted*

### DETAILED STEPS

## Examples

**DETAILED STEPS**

## Examples

The following example sets a timeout of 20 seconds for calls that are transferred to busy destinations:

```
call-manager-fallback
 timeouts busy 20
```

# Configuring the Ringing Timeout Default

The ringing timeout default is the length of time for which a phone can ring with no answer before

### DETAILED STEPS

## Examples

The following example sets the ringing timeout default to 30 seconds:

```
call-manager-fallback
 timeouts ringing 30
```

# Configuring Outgoing Calls

Outgoing call configuration can include the following tasks:

- Configuring Call Transfer
  - Configuring Local and Remote Call Transfer, page 123 (Optional)
  - Enabling Consultative Call Trl–Chaptere Calm.92 31(2)386 CoJ/F5..11F5.0J0a0 TD[( 3(1/F5.24.7(5(44

### SUMMARY STEPS

1. **call-manager-fallback**
2. **transfer-pattern** *transfer-pattern*
3. **exit**

### DETAILED STEPS

## Examples

## Examples

The following example enables the H.450 Tcl script for analog transfer using hookflash and sets a delay time of 1 second:

```
call application voice transfer_app flash:app-h450-transfer.tcl
call application voice transfer_app language 1 en
call application voice transfer_app set-location en 0 flash:/prompts
call application voice transfer_app delay-time 1
!
dial-peer voice 25 pots
 destination-pattern 9.T
 port 1/0/0
 application transfer_app
!
dial-peer voice 29 voip
 destination-pattern 4…
 session-target ipv4:10.1.10.1
 application transfer_app
```

## Configuring Trunk Access Codes

## DETAILED STEPS

## Examples

The following example shows how to set a dial-peer COR parameter for outgoing calls to the Cisco Unified IP Phone dial peers and directory numbers created during fallback:

```
call-manager-fallback
 cor outgoing LockforPhoneC 1 5010 - 5020
```

10. **codec** *codec-type* [*bytes*]
11. **end**

## DETAILED STEPS

# Configuring SIP-to-SIP Call Forwarding

SIP-to-SIP call forwarding (call routing) is available. Call forwarding is provided either by the phone or by using a back-to-back user agent (B2BUA), which allows call forwarding on any dial peer. Calls into

CHAPTER

### SIP SRST

- Cisco 4000 Series Integrated Services Router only supports Secure SIP SRST. The router series does not support Secure SCCP SRST.
- SRTP passthrough is not supported.
- supp188.4(R)-4.8(ST)]T cong in1.5d1 TD.021not .68 2F

## SRST Routers and the TLS Protocol

Transport Layer Security (TLS) Version 1.0 provides

## Cisco Unified Communications Manager 4.X.X and Earlier Versions

For systems running Cisco Unified Communications Manager 4.X.X and earlier versions, the secure Cisco Unified SRST Router must retrieve phone certificates so that it can authenticate Cisco Unified IP phones during the TLS handshake. Different certificates are used for different Cisco Unified IP Phones. Table 8-1

For complete information on adding Cisco Unified SRST to Cisco Unified Communications Manager,

| Command or Action | Purpose |
|---|---|
| `button 1: dn 21 number 2011 CM Fallback CH1`<br>`CONNECTED`<br>`Active Secure Call on DN 21 chan 1 :2011 10.1.1.40`<br>`16382 to 10.1.1.40 16382 via 10.1.1.40`<br>`G711Ulaw64k 160 bytes no vad`<br>`Tx Pkts 295 bytes 49468 Rx Pkts 277 bytes 46531 Lost`<br>`0`<br>`Jitter 0 Latency 0 callingDn -1 calledDn 11` | |

## Configuration Examples for Secure SCCP SRST

This section provides the following configuration examples:

- Secure SCCP SRST: Example, page 197
- Control Plane Policing: Example, page 202

```
  5EE85FF8 C1B1A540 E818CE6D 58131726 BB060974 4E1A2F4B E6195522 122457F3
  DEDBAAD7 3780136E B112A6
  quit
crypto pki certificate chain srstcaserver
 certificate ca 01
  30820207 30820170 A0030201 02020101 300D0609 2A864886 F70D0101 04050030
  17311530 13060355 0403130C 73727374 63617365 72766572 301E170D 30343034
  31323139 34353136 5A170D30 37303431 32313934 3531365A 30173115 30130603
  55040313 0C737273 74636173 65727665 7230819F 300D0609 2A864886 F70D0101
  01050003 818D0030 81890281 8100C3AF EE1E4BB1 9922A8DA 2BB9DC8E 5B1BD332
  1051C9FE 32A971B3 3C336635 74691954 98E765B1 059E24B6 32154E99 105CA989
  9619993F CC72C525 7357EBAC E6335A32 2AAF9391 99325BFD 9B8355EB C10F8963
  9D8FC222 EE8AC831 71ACD3A7 4E918A8F D5775159 76FBF499 5AD0849D CAA41417
  DD866902 21E5DD03 C37D4B28 0FAB0203 010001A3 63306130 0F060355 1D130101
  FF040530 030101FF 300E0603 551D0F01 01FF0404 03020186 301D0603 551D0E04
  160414F8 29CE97AD 6018D054 67FC2939 63C24706 91F9BD30 1F060355 1D230418
  30168014 F829CE97 AD6018D0 5467FC29 3963C247 0691F9BD 300D0609 2A864886
  F70D0101 0405000  1 0355 1D230418 3.5(6792C805 29CE97AD 6047A810 5467F0 -95B5AAE51D6 B0BB7.559TJ
  DEDBAAD
  quit
```

### DETAILED STEPS

## Configuring SIP options for Secure SIP SRST

This section explains how to configure secure SIP SRTP.

### SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **url sip | sips**
6.

```
!
crypto pki trustpoint Cisco_Root_CA_2048
 enrollment terminal
 revocation-check none
!
!
crypto pki certificate chain TRUSTPT-SRST-CA-2
 certificate 02
  3082020B 30820174 A0030201 02020102 300D0609 2A864886 F70D0101 05050030
  14311230 10060355 04031309 53525354 2D43412D 32301E17 0D313730 36303831
  31333131 325A1714.029E3036 30383131 33313132 5A303231 30301206 03550405
  130B4647 4C313735 31313150 42301A06 092A8648 86F70D01 09021614.416E7473
  41726D79 2D343430 3030819F 300D0609 2A864886 F70D0101 01050003 818D0030
  818AE28 0A(050 .6 824 6259A98D A61C01)7.5947345A95DA8 DE83ECAD C201B4648741F7E.642
  D753BF8 0A(0519BD54FB 9A4D4A8E 7A2BA8 0A(.41 B93C40B3 A63A7C4D 723)7.5(3498F1 08EF07F3 )]TJ T* [
  37456F8 0A(36 D20387C D46333FA469FB20)7.59E81 311E01C A7AB19A3 964
 A08D70D710003D01.00A3 4F3064205A30 31F00603
```

## Related Documents

## Standards

## MIBs

## RFCs

# Feature Information for Secure SCCP and SIP SRST

Table 8-4 lists the release history for this feature.

Not all commands may be available in your Cisco IOS software release. For release information about a specific command, see the command reference documentation.

Use Cisco Feature Navigator to find information about platform support and software image support.

- Hunt Groups

The list of SIP trunk features supported for Unifie

- Avoid configuring dial-peer groups on the SIP trunk dial-peer pointing to the Service Provider router.
- Configure the destination pattern (.T) on the dial-peer that points to Unified Communications Manager.

```
voice-class sip profiles 201
```

## Examples

*Figure 10-3 How Voicemail Dial*

CHAPTER

```
preference 0 secondary 9
huntstop
call-waiting beep

ephone-dn 10
preference 0 secondary 9
huntstop
call-waiting beep

ephone-dn 11
preference 0 secondary 9
huntstop
call-waiting beep

ephone-dn 12
preference 0 secondary 9
huntstop
call-waiting beep

ephone-dn 13
preference 0 secondary 9
huntstop
call-waiting beep

ephone-dn 14
preference 0 secondary 9
huntstop
call-waiting beep

ephone-dn 15
preference 0 secondary 9
huntstop
call-waiting beep

ephone-dn 16
preference 0 secondary 9
huntstop
call-waiting beep

ephone-dn 17
preference 0 secondary 9
huntstop
call-waiting beep

ephone-dn 18
preference 0 secondary 9
huntstop
call-waiting beep

ephone-dn 19
preference 0 secondary 9
huntstop
call-waiting beep

ephone-dn 20
preference 0 secondary 9
huntstop
call-waiting beep

Number of Configured ephones 0 (Registered 2)

voice-port 50/0/1
 station-id number 1001
 station-id name 1001
```

```
 port 50/0/16

dial-peer voice 20071 pots
 huntstop
 progress_ind setup enable 3
 port 50/0/17

dial-peer voice 20072 pots
 huntstop
 progress_ind setup enable 3
 port 50/0/18

dial-peer voice 20073 pots
 huntstop
 progress_ind setup enable 3
 port 50/0/19

dial-peer voice 20074 pots
 huntstop
 progress_ind setup enable 3
 port 50/0/20

tftp-server system:/its/SEPDEFAULT.cnf
tftp-server system:/its/SEPDEFAULT.cnf alias SEPDefault.cnf
tftp-server system:/its/XMLDefault.cnf.xml alias XMLDefault.cnf.xml
tftp-server system:/its/ATADefault.cnf.xml
tftp-server system:/its/united_states/7960-tones.xml alias United_States/7960-tones.xml
tftp-server system:/its/united_states/7960-font.xml alias
English_United_States/7960-font.xml
tftp-server system:/its/united_states/7960-dictionary.xml alias
English_United_States/7960-dictionary.xml
tftp-server system:/its/united_states/7960-kate.xml alias
English_United_States/7960-kate.xml
tftp-server system:/its/united_states/SCCP-dictionary.xml alias
English_United_States/SCCP-dictionary.xml
```

| | |
|---|---|
| `Router# show running-config` | Displays the configuration. |

–

## Verifying Cisco Unified SRST MOH Live Feed