Cisco Unified Survivable Remote Site Telephony Feature Roadmap

This chapter contains a list of Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) features and the location of feature documentation.

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at http://www.cisco.com/go/fn. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

Documentation Organization

This document consists of the following chapters or appendixes as shown in the following table .

Chapter or Appendix	Description
Cisco Unified SRST Feature Overview	Gives a brief description of Cisco Unified SRST and provides information on the supported platforms and Cisco Unified IP Phones. In addition, it describes any prerequisites or restrictions that should be addressed before Cisco Unified SIP SRST is configured.
Setting Up the Network	Describes how to set up a Cisco Unified SRST system to communicate with your network.
Cisco Unified Enhanced Survivable Remote Site Telephony	Describes how to configure the Cisco Unified Enhanced SRST feature in your network.
Cisco Unified SIP SRST 4.1	Describes the features for Cisco Unified SIP SRST Version 4.1 and provides the associated configuration procedures.
Setting Up Cisco Unified IP Phones using SCCP	Describes how to set up the basic Cisco Unified SRST phone configuration.
Setting Up Cisco Unified IP Phones using SIP	Describes features available in Version 3.0 that are also necessary for Version 3.4. Features include instructions on how to provide a backup to an external SIP call control (IP-PBX) by providing basic registrar services. These services are used by a SIP IP phone in the event of a WAN connection outage when the SIP phone is unable to communicate with its primary SIP proxy.
Configuring Call Handling	Describes how to configure incoming and outgoing calls.
Configuring Secure SRST for SCCP and SIP	Describes the Secure SRST security functionality to the Cisco Unified SRST.
Integrating Voicemail with Cisco Unified SRST	Describes how to set up voicemail.
Setting Video Parameters	Describes how to set up video parameters.
Monitoring and Maintaining Cisco Unified SRST	Provides a list of useful show commands for monitoring and maintaining Cisco Unified SRST.
Configuring Cisco Unified SIP SRST Features Using Redirect Mode	Describes features using redirect mode, which applies to version 3.0 only.

Chapter or Appendix	Description
Integrating Cisco Unified Communications Manager and Cisco Unified SRST to Use Cisco Unified SRST as a Multicast MOH Resource	Describes how to configure Cisco Unified CM and Cisco Unified SRST to enable multicast music-on-hold (MOH).

Feature Roadmap

The following table provides a feature history summary of Cisco Unified SRST features.

Cisco Unified SRST	Cisco IOS Release	Enhancements or Modifications
Version 14.1	Cisco IOS XE Bengaluru 17.5.1a Release	• Support for Unified SRST on Cisco 1100 Integrated Services Router. • Support for Unified SRST on Cisco 8200L Catalyst Edge Series Platform.
Version 14.1	Cisco IOS XE Bengaluru 17.4.1a Release	• Support for Unified SRST on Cisco 8200 Catalyst Edge Series Platform. • Smart Licensing Using Policy—Cisco Smart Licensing for Unified SRST • Smart Licensing Using Policy—Cisco Smart Licensing for Unified E-SRST
Version 14.1	Cisco IOS XE Amsterdam 17.3.2 Release	• Support for Unified SRST on Cisco 8300 Catalyst Edge Series Platforms • Smart Licensing Using Policy—Cisco Smart Licensing for Unified SRST • Smart Licensing Using Policy—Cisco Smart Licensing for Unified E-SRST
Version 12.8	Cisco IOS XE Amsterdam 17.2.1r	• Cisco Jabber with Unified SRST, page 34 • VRF Support for Unified SRST • Support for YANG Models in Unified SRST
Version 12.7	Cisco IOS XE Amsterdam 17.1.1	Support for maximum number of devices in Cisco 4451 and 4461 Integrated Services Routers was increased from 1500 to 2000.

Cisco Unified SRST	Cisco IOS Release	Enhancements or Modifications
Version 12.6	Cisco IOS XE Gibraltar 16.11.1a	• Simple Network Management Protocol (SNMP) Support for Unified SRST • Toll Fraud Prevention for SIP Line Side on Unified SRST • Unified SRST, Unified E-SRST, and Unified Secure SRST Password Policy
Version 12.5	Cisco IOS XE Gibraltar 16.10.1a	Support for Unified SRST on Cisco 4461 Integrated Services Routers
Version 12.3	Cisco IOS XE Fuji 16.9.1	Secure SCCP SRST on Cisco 4000 Series Integrated Services Router
Version 12.2	Cisco IOS XE Fuji 16.8.1	Unified E-SRST with Support for Voice Hunt Group
Version 12.1	Cisco IOS XE Fuji 16.7.1	• Licensing • Secure SCCP SRST on Cisco 4000 Series Integrated Services Router • Unified SRST and Unified Border Element Co-location
Version 12.0	Cisco IOS XE Everest 16.6.1	IPv6 Support for Unified SRST SIP IP Phones
Version 11.0	15.6(1)T	• Support for Cisco IP Phone 7811 • Support for Cisco IP Phones 8811, 8831, 8841, 8845, 8865, 8851, 8851NR, 8861 • Support for Cisco ATA-190 Phones
Version 10.5	15.4(3)M	• Setting up the Network • Support for Cisco Unified DX650 SIP IP Phones • Support for Cisco Unified 78xx SIP IP Phones • Support for Cisco IP Phones 88xx, 8941, 8945, and 8961
Version 10.0	15.3(3)M	• Cisco Jabber for Windows • SIP: Configure Unified E-SRST

Cisco Unified SRST	Cisco IOS Release	Enhancements or Modifications
Version 9.5	15.3(2)T	• After-hour Pattern Blocking Support for Regular Expressions • Call Park Recall Enhancement • Display Support for Name of Called Voice Hunt Groups • Preventing Local-Call Forwarding to Final Agent in Voice Hunt Groups • Trunk-to-Trunk Transfer Blocking for Toll Fraud Prevention on Cisco Unified SIP IP Phones
Version 9.1	15.2(4)M	• Key Expansion Module Support for Cisco Unified SIP IP Phones • Enhancement in Speed-Dial Support • Voice Hunt Group Support
Version 9.0	15.2(2)T	• Support for Cisco Unified 6901 and 6911 SIP IP Phones • Support for Cisco Unified 6921, 6941, 6945, and 6961 SIP IP Phones • Support for Cisco Unified 8941 and 8945 SIP IP Phones • Multiple Calls Per Line • Voice and Fax Support on Cisco ATA-187
Version 8.8	15.2(1)T	Support for Cisco Unified 6945, 8941, and 8945 SCCP IP Phones
Version 8.6	15.1(4)M	Support for Cisco Unified 8941 and 8945 SCCP IP Phones were introduced. For more information, see Configuring Cisco Unified 8941 and 8945 SCCP IP Phones.
Version 8.0	15.1(1)T	Beginning with Cisco IP Phone firmware 8.5(3) and Cisco IOS Release 15.1(1)T, Cisco SRST supports SIP signaling over UDP, TCP, and TLS connections, providing both RTP and SRTP media connections based on the security settings of the IP phone. For more information, see the following sections: • Signaling Security on Unified SRST - TLS • Media Security on Unified SRST - SRTP • Configuring Secure SIP Call Signaling and SRTP Media with Cisco SRST
Version 7.0/4.3	See Cisco Feature Navigator for compatibility.	• Configuring Eight Calls per Button (Octo-Line) • Configuring Consultative Transfer

Cisco Unified SRST	Cisco IOS Release	Enhancements or Modifications
Version 4.2(1)	See Cisco Feature Navigator for compatibility.	Enhanced 911 Services. The following new features are included: • Assigning ERLs to zones to enable routing to the PSAP that is closest to the caller. • Customizing E911 by defining a default ELIN, identifying a designated number if the 911 caller cannot be reached on callback, specifying the expiry time for data in the Last Caller table, and enabling syslog messages that announce all emergency calls. • Expanding the E911 location information to include name and address. • Adding new permanent call detail records.
Version 4.1	12.4(15)T	• Enabling KPML for SIP Phones • Disabling SIP Supplementary Services for Call Forward and Call Transfer • Configuring idle Prompt Status for SIP Phones • Enhanced 911 Services
Version 4.0	12.4(4)XC	• Cisco IP Communicator Support • Fax Pass-through using SCCP and ATAs Support • H.323 VoIP Call Preservation Enhancements for WAN Link Failures for SCCP Phones • Video Support
Version 3.4	12.4(4)T	• Cisco SIP SRST 3.4 • Configuring Cisco Unified SIP SRST Features Using Redirect Mode • Configuring Call Handling (see Back-to-Back User Agent Mode)
Version 3.3		• Secure SRST • Cisco Unified IP Phone 7970G and Cisco Unified 7971G-GE Support • Enhancement to the show ephone Command

Cisco Unified SRST	Cisco IOS Release	Enhancements or Modifications
Version 3.2	12.3(11)T	• Enhancement to the alias Command • Enhancement to the pickup Command • Enhancement to the user-locale Command • Increased the Number of Cisco Unified IP Phones Supported on the Cisco 3845 • MOH Live-Feed Support • No Timeout for Call Preservation • RFC 2833 DTMF Relay Support • Translation Profile Support
Version 3.1	12.3(7)T	• Cisco Unified IP Phone 7920 Support • Cisco Unified IP Phone 7936 Support
Version 3.0	12.2(15)ZJ 12.3(4)T	• Additional Language Options for IP Phone Display • Consultative Call Transfer and Forward Using H.450.2 and H.450.3 for SCCP Phones • Customized System Message for Cisco Unified IP Phones • Dual-Line Mode • E1 R2 Signaling Support • European Date Formats • Huntstop for Dual-Line Mode • Music On Hold for Multicast from Flash Files • Ringing Timeout Default • Secondary Dial Tone • Enhancement to the Show ephone Command • System Log Messages for Phone Registrations • Three-Party G.711 Ad Hoc Conferencing • Support for Cisco VG248 Analog Phone Gateway 1.2(1) and Higher Versions

Cisco Unified SRST	Cisco IOS Release	Enhancements or Modifications
Version 2.1		• Cisco Unified IP Phone 7902G Support • Cisco Unified IP Phone 7912G Support • Additional Language Options for IP Phone Display • Cisco Unified SRST Aggregation • Cisco ATA 186 and ATA 188 Support • Cisco Unified IP Phone 7905G Support • Cisco Unified IP Phone Expansion Module 7914 Support • Enhancement to the Dial Plan-Pattern Command
Version 2.02		• Cisco Unified IP Phone Conference Station 7935 Support • Increase in Directory Numbers • Cisco Unity Voicemail Integration Using In-Band DTMF Signaling Across the PSTN and BRI/PRI, on page 42 • Cisco Unified SRST was implemented on the Cisco Catalyst 4500 access gateway module and Cisco 7200 routers (NPE-225, NPE-300, and NPE400). • Support was removed for the Cisco MC3810-V3 concentrator.
Version 2.01		• Cisco Unified SRST was implemented on the Cisco 1760 routers, and support for the Cisco 1750 was removed. • Support was added for additional connected Cisco IP phones. • Support was added for additional directory numbers or virtual voice ports on Cisco IP phones.

Cisco Unified SRST	Cisco IOS Release	Enhancements or Modifications
Version 2.0		• Cisco Unified SRST was implemented on the Cisco 2600XM and Cisco 2691 routers. • Cisco Unified SRST was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers and the Cisco MC3810-V3 concentrators. • Cisco Unified SRST was implemented on the Cisco 1750 and Cisco 1751 routers. • Huntstop support. • Class of restriction (COR). • Translation rule support. • MOH and tone on hold. • Distinctive ringing. • Forward to a central voicemail or auto-attendant (AA) through PSTN during Cisco Unified Communications Manager fallback. • Phone number alias support during Cisco Unified Communications Manager fallback: enhanced default destination support. • List-based call restrictions for Cisco Unified Communications Manager fallback.

Cisco Unified SRST	Cisco IOS Release	Enhancements or Modifications
Version 1.0		• Support was added for 144 Cisco IP phones on the Cisco 3660 multiservice routers. • Cisco Unified SRST introduced on the Cisco 2600 series and Cisco 3600 series multiservice routers and the Cisco IAD2420 series integrated access devices. • Cisco IP phones able to establish a connection with an SRST router in the event of a WAN link to Cisco Unified Communications Manager failure. • Dimming of all Cisco Unified IP Phone function keys that are not supported during Cisco Unified SRST operation. • Extension-to-extension dialing. • Direct Inward Dialing (DID). • Direct Outward Dialing (DOD). • Calling party ID (Caller ID/ANI) display. • Last number redial. • Preservation of local extension-to-extension calls when WAN link fails. • Preservation of local extension to PSTN calls when WAN link fails. • Preservation of calls in progress when failed WAN link is re-established. • Blind transfer of calls within IP network. • Multiple lines per Cisco IP phone. • Multiple-line appearance across telephones. • Call hold (shared lines). • Analog Foreign Exchange Station (FXS) and Foreign Exchange Office (FXO) ports. • BRI support for EuroISDN. • PRI support for NET5 switch type.

Information About New Features in Cisco Unified SRST

This section contains the following topics:

- New Features for Unified SRST Version 14.1

New Features for Unified SRST Version 14.1

Unified SRST 14.1 Release introduces support for the following new features:

- Smart Licensing Using Policy—Cisco Smart Licensing for Unified SRST
- Smart Licensing Using Policy—Cisco Smart Licensing for Unified E-SRST

New Features for Unified SRST Version 12.7

Unified SRST 12.7 Release introduces support for the following new feature:

- Support for maximum number of devices in Cisco 4451 and 4461 Integrated Services Routers was increased from 1500 to 2000.

New Features for Cisco Unified SRST Version 12.6

Cisco Unified SRST 12.6 Release introduces support for the following new features:

- Simple Network Management Protocol (SNMP) Support for Cisco Unified SRST
- Toll Fraud Prevention for SIP Line Side on Cisco Unified SRST
- Cisco Unified SRST, Unified E-SRST, and Unified Secure SRST Password Policy

New Features for Cisco Unified SRST Version 12.3

Cisco Unified SRST 12.3 Release introduces support for Secure SCCP SRST on Cisco 4000 Series Integrated Services Router.

New Features for Cisco Unified SRST Version 12.2

Cisco Unified SRST 12.2 Release introduces support for Unified E-SRST with Support for Voice Hunt Group.

New Features for Cisco Unified SRST Version 12.1

Cisco Unified SRST 12.1 introduces support for the following new features:

- Licensing
- Secure SIP SRST Support on Cisco 4000 Series Integrated Services Router
- Cisco Unified SRST and Cisco Unified Border Element co-location

New Feature for Cisco Unified SRST Version 12.0

Cisco Unified SRST 12.0 introduces support for IPv6 protocols on SIP IP Phones. For more information on IPv6 Support introduced for Cisco Unified SRST, see IPv6 Support for Cisco Unified SRST SIP IP Phones.

New Features for Cisco Unified SRST Version 11.0

Cisco Unified SRST 11.0 supports the following new Cisco IP phones and adapters:

- Support for Cisco IP Phone 7811
- Support for Cisco IP Phones 8811, 8831, 841, 8851, 8851NR, 8861
- Support for Cisco ATA-190

For information on the phones supported in Cisco Unified SRST 11.0, see Phone Feature Support Guide for Cisco Unified Communications Manager Express, Cisco Unified SRST, Unified E-SRST, and Unified Secure SRST.

New Features for Cisco Unified SRST Version 10.5

Cisco Unified SRST 10.5 supports the following features:

- Where to Go Next, Setting Up the Network

For more information on the Cisco Unified SRST 10.5 supported feature, see the SCCP: Configure Unified E-SRST.

Cisco Unified SRST 10.5 supports the following new Cisco Unified SIP IP phones:

- Support for Cisco Unified DX650 SIP IP Phones
- Support for Cisco Unified 78xx SIP IP Phones

Support for Cisco Unified DX650 SIP IP Phones

For information on feature support for the Cisco Unified DX650 SIP IP Phones in Cisco Unified SRST 10.5, see Phone Feature Support Guide for Unified CME, Unified SRST, Unified E-SRST, and Unified Secure SRST.

Support for Cisco Unified 78xx SIP IP Phones

For information on feature support for the Cisco Unified 78xx SIP IP Phones in Cisco Unified SRST 10.5, see Phone Feature Support Guide for Unified CME, Unified SRST, Unified E-SRST, and Unified Secure SRST.

New Features in Cisco Unified SRST Version 10.0

Cisco Unified SRST 10.0 supports the following new features:

- Cisco Jabber for Windows
- SIP: Configure Unified E-SRST

To obtain an account on Cisco.com, go to www.cisco.com and click Register at the top of the screen.

Cisco Jabber for Windows

Cisco Jabber for Windows client is supported from Cisco Unified CME Release 10 onwards.Cisco Jabber for Windows supports the visual voicemail functionality integrated with the Cisco Unity connection. Cisco Jabber for Windows is a SIP-based soft client with integrated Instant Messaging and presence functionality, and uses the new Client Services Framework 2nd Generation (CSF2G) architecture.

CSF is a unified communications engine that is reused by multiple Cisco PC-based clients. The Cisco Jabber client has to be registered with a presence server such as cloud-based Cisco Webex server, or Cisco Unified Presence server to avail the standard XMPP-based instant messaging functionalities. The client is identified by a device ID name that can be configured under the voice register pool in Cisco Unified CME. You should configure the username and password under voice register pool to identify the user logging into Cisco Unified CME through Cisco Jabber for Windows client. The device discovery process uses HTTPS connection. Therefore, you should configure the secure HTTP on Cisco Unified CME. A new phone type, Jabber-Win has been added to configure the voice register pool for Cisco Jabber for Windows client.

Restrictions

- The Cisco Jabber for Windows client version should be version 9.1.0 and later version.
- The Cisco Jabber for Windows client should register with a presence server such as cloud-based Webex server, or a Cisco Unified Presence server to enable the telephony features on the Jabber client.
- The Cisco Jabber for Windows client supports only the visual voicemail functionality using Internet Message Access Protocol (IMAP) on the Cisco Unity Connection.
- The Cisco Jabber for Windows client does not support software-based conferencing and supports only the softphone mode with Cisco Unified CME.
- Desk phone models are not supported.

For configuration information, see the “Cisco Jabber for Windows” section of Cisco Unified Communications Manager Administration Guide.

Version Negotiation for Cisco Unified SIP IP Phones

The version negotiation for Cisco Unified SIP IP Phones was introduced in Cisco Unified SRST 10.0 release. For more information on the Cisco Unified SRST 10.0 supported features, see the SIP: Configure Unified E-SRST section.

New Features in Cisco Unified SRST Version 9.5

After-hour Pattern Blocking Support for Regular Expressions

In Cisco Unified SRST 9.5, support for afterhours pattern blocking is extended to regular expression patterns for dial plans on Cisco Unified SIP and Cisco Unified SCCP IP phones. With this support, users can add a combination of fixed dial plans and regular expression-based dial plans.

When a call is initiated after hours, the dialed number is matched against a combination of dial plans. If a match is found, the call is blocked.

To enable regular expression patterns to be included when configuring afterhours pattern blocking, the **after-hours block pattern** command is modified to include regular expressions as a value for the *pattern* argument in the following command syntax:

after-hours block pattern *pattern-tag pattern*

This command is available in the following configuration modes:

- telephony-service—For both SCCP and SIP Phones.
- ephone-template—For SCCP phones only.

Note The maximum length of a regular expression pattern is 32 for both Cisco Unified SIP and Cisco Unified SCCP IP phones.

If calls to the following numbers are to be blocked after hours:

- numbers beginning with ‘0’ and ‘00’
- numbers beginning with 1800, followed by four digits
- numbers 9876512340 to 9876512345

then the following configurations can be used:

- after-hours block pattern 1 0*
- after-hours block pattern 2 00*
- after-hours block pattern 3 1800….
- after-hours block pattern 4 987651234[0-5]

Note There is no change in the number of afterhours patterns that can be added. The maximum number is still 100.

For more information on configuration examples, see the “Configuring Afterhours Block Patterns of Regular Expressions: Example” section of Cisco Unified Communications Manager Administration Guide.

For a summary of the basic Cisco IOS regular expression characters and their functions, see the Cisco Regular Expression Pattern Matching Characters section of *Terminal Services Configuration Guide*.

Call Park Recall Enhancement

Before Cisco Unified CME 9.5, a parked call could not be recalled by or transferred to the phone that put the call in park or the original phone that transferred the call when the destination phone was offhook or ringing.

In Cisco Unified CME 9.5, the **recall force** keyword is added to the **call-park system** command in telephony-service configuration mode to allow a user to force the recall or transfer of a parked call to the phone that put the call in park or the phone with the reserved-for number as its primary DN when the destination phone is available to answer the call.

In Cisco Unified CME 10.5, a new ring tone is introduced for park recall to assist the phone user to distinctly identify the type of call.

This feature is supported on all phone families for SCCP endpoints and on 89XX and 99XX phone families for SIP endpoints. No configurations are required to activate this feature.

The following example configures the Call Park Recall:

```
Router# configure terminal
Router(config)# telephony-service
Router(config)# srst mode auto-provision all
Router(config-telephony)# call-park system ? recall Configure parameters for recall
Router(config-telephony)# call-park system recall ? force Force recall for busy call park
initiator
Router(config-telephony)# call-park system recall force
```

Park Monitor

In Cisco Unified CME 8.5 and later versions, the park monitor feature allows you to park a call and monitor the status of the parked call until the parked call is retrieved or abandoned. When a Cisco Unified SIP IP Phone 8961, 9951, or 9971 parks a call using the park soft key, the park monitoring feature monitors the status of the parked call. The park monitoring call bubble is not cleared until the parked call gets retrieved or is abandoned by the parkee. This parked call can be retrieved using the same call bubble on the parker's phone to monitor the status of the parked call.

Once a call is parked, Cisco Unified CME sends a SIP NOTIFY message to the parker phone indicating the "parked" event along with the park slot number so that the parker phone can display the park slot number as long as the call remains parked.

When a parked call is retrieved, Cisco Unified CME sends another SIP NOTIFY message to the parker phone indicating the "retrieved" event so that the phone can clear the call bubble. When a parked call is disconnected by the parkee, Cisco Unified CME sends a SIP NOTIFY message to the parker phone indicating the "abandoned" event and the parker phone clears the call bubble upon cancellation of the parked call.

When a parked call is recalled or transferred, Cisco Unified CME sends a SIP NOTIFY message to the parker phone indicating the "forwarded" event so that parker phone can clear the call bubble during park, recall, and transfer. You can also retrieve a parked call from the parker phone by directly selecting the call bubble or pressing the resume soft key on the phone.

Display Support for Name of Called Voice Hunt Groups

A voice hunt group is associated with a pilot number. But because there is no association with the name of the voice hunt group when calls are forwarded from the voice hunt group to the final number, the forwarding number is sent without the name of the forwarding party. The final number can be in the form of a voicemail, a Basic Automatic Call Distribution (BACD) script, or another extension.

In Cisco Unified SRST 9.5, the display of the name of the called voice-hunt-group pilot is supported by configuring the following command in **voice hunt-group** or **ephone-hunt** configuration mode:

[**no**] **name** *"primary pilot name"* [**secondary** *"secondary pilot name"*]

The secondary name is optional and when the secondary pilot name is not explicitly configured, the primary pilot name is applicable to both pilot numbers.

For configuration information, see the "Associating a Name with a Called Voice Hunt Group" section of Cisco Unified Communications Manager Administration Guide.

For configuration examples, see the "Example: Associating a Name with a Called Voice Hunt Group" section of Cisco Unified Communications Manager Administration Guide.

Restrictions

- Display support applies to Cisco Unified SCCP IP phones in voice hunt-group and ephone-hunt configuration modes but are not supported in Cisco Unified SIP IP phones.
- Called name and called number information displayed on the caller's phone follows existing behavior, where the called names and called numbers are updated so that a sequential hunt reflects the name and number of the ringing phone.

The following example configures the primary pilot name for both the primary and secondary pilot numbers:

```
name SALES
```

The following example configures different names for the primary and secondary pilot numbers:

```
name SALES secondary SALES-SECONDARY
```

Note Use quotes (") when input strings have spaces in between as shown in the next three examples.

The following example associates a two-word name for the primary pilot number and a one-word name for the secondary pilot number:

```
name “CUSTOMER SERVICE” secondary CS
```

The following example associates a one-word name for the primary pilot number and a two-word name for the secondary pilot number:

```
name FINANCE secondary “INTERNAL ACCOUNTING”
```

The following example associates two-word names for the primary and secondary pilot numbers:

```
name “INTERNAL LLER” secondary “EXTERNAL LLER”
```

Preventing Local-Call Forwarding to Final Agent in Voice Hunt Groups

Local or internal calls are calls originating from a Cisco Unified SIP or Cisco Unified SCCP IP phone in the same Cisco Unified CME system.

Before Cisco Unified CME 9.5, the **no forward local-calls** command was configured in ephone-hunt group to prevent a local call from being forwarded to the next agent.

In Cisco Unified CME 9.5, local calls are prevented from being forwarded to the final destination using the **no forward local-calls to-final** command in parallel or sequential voice hunt-group configuration mode.

When the **no forward local-calls to-final** command is configured in sequential voice hunt-group configuration mode, local calls to the hunt-group pilot number are sent sequentially only to the list of members of the group using the rotary-hunt technique. In case all the group members of the voice hunt group are busy, the caller hears a busy tone. If any of the group members are available but do not answer, the caller hears a ringback tone and is eventually disconnected after the specified timeout. The call is not forwarded to the final number.

When the **no forward local-calls to-final** command is configured in parallel voice hunt-group configuration mode, local calls to the hunt-group pilot number are sent simultaneously to the list of members of the group using the blast technique. In case all the group members of the voice hunt group are busy, the caller hears a busy tone. If any of the group members are available but do not answer, the caller hears a ringback tone and is eventually disconnected after the specified timeout.The call is not forwarded to the final number. or configuration examples, see the "Preventing Local-Call Forwarding to Final Agent in Voice Hunt Groups" section of" section of Cisco Unified Communications Manager Administration Guide.

Trunk-to-Trunk Transfer Blocking for Toll Fraud Prevention on Cisco Unified SIP IP Phones

In Cisco Unified Survivable Remote Site Telephony (SRST) 4.0, trunk-to-trunk transfer blocking for toll bypass fraud prevention is supported on Cisco Unified Skinny Client Control Protocol (SCCP) IP phones.

The following table lists the transfer-blocking commands and the appropriate configuration modes for Cisco Unified CME and Cisco Unified SRST.

Commands	Cisco Unified SRST
transfer-pattern	call-manager-fallback
transfer max-length	voice register pool
transfer-pattern blocked	voice register pool
conference transfer-pattern	call-manager-fallback
conference max-length	voice register pool or voice register template
conference-pattern blocked	voice register pool or voice register template

Note The call transfer and conference restrictions apply when transfers or conferences are initiated toward external parties, like a PSTN trunk, a SIP trunk, or an H.323 trunk. The restrictions do not apply to transfers and conferences to local extensions.

Transfer-Pattern

The **transfer-pattern** command for Cisco Unified SIP IP phones functions like the **transfer-pattern** command for Cisco Unified SCCP IP phones by allowing all, not just local, transfers to take place.

The **transfer-pattern** command specifies the directory numbers for Call Transfer. The command can be configured up to 32 times using the following command syntax: **transfer-pattern** *transfer-pattern* [**blind**].

Note The **blind** keyword in the **transfer-pattern** command applies only to Cisco Unified SCCP IP phones and does not apply to Cisco Unified SIP IP phones.

With the **transfer-pattern** command configured, only Call Transfers to numbers that match the configured transfer pattern are allowed to take place. With the transfer pattern configured, all or a subset of transfer numbers can be dialed and the transfer to a remote party can be initiated.

The following are examples of configurable transfer patterns:

- .T—This configuration allows Call Transfers to any destinations with one or more digits, like 123, 877656, or 76548765.
- 919........—This configuration only allows Call Transfers to remote numbers beginning with “919” and followed by eight digits, like 91912345678. However, Call Transfers to 9191234 or 919123456789 are not allowed.

Backward Compatibility

To maintain backward compatibility, all Call Transfers from Cisco Unified SIP IP phones to any number (local or over the trunk) are allowed when no transfer patterns are configured through the **transfer-pattern** , **transfer-pattern blocked**, or **transfer max-length** commands.

For Cisco Unified SCCP IP phones, if you do not configure transfer patterns, Call Transfers over the trunk are blocked.

Dial Plans

Whatever dial plan is used for external calls, the same numbers should be configured as specific numbers using the**transfer-pattern** command.

If a dial plan requires “9” to be dialed before making an external call, then prefix “9” to the transfer-pattern number. For example, if 12345678 is an external number that requires “9” to be dialed before making the external call, then the transfer-pattern number is 912345678.

Transfer Max-Length

The **transfer max-length** command is used to indicate the maximum length of the number being dialed for Call Transfer. When only a specific number of digits are allowed during a Call Transfer, value from 3 through 16 is configured. When the number dialed exceeds the maximum length, then the Call Transfer is blocked.

For example, if you configure 5 as the maximum length, Call Transfers from Cisco Unified SIP IP phones allows up to a five-digit directory number. All Call Transfers to directory numbers with more than five digits are blocked.

Note If only **transfer max length** is configured and **conference max-length** is not configured, then **transfer max length** takes effect for transfers and conferences.

Transfer-Pattern Blocked

When the **transfer-pattern blocked** command is configured for a specific phone, no Call Transfers are allowed from that phone over the trunk.

This feature forces unconditional blocking of all Call Transfers from the specific phone to any other nonlocal numbers (external calls from one trunk to another trunk). No Call Transfers from this specific phone are possible even when a transfer pattern matches the dialed digits for transfer.

The following table compares the behaviors of Cisco Unified SCCP and SIP IP phones for specific configurations.

Configuration	Cisco Unified SCCP IP Phones	Cisco Unified SIP IP Phones
No transfer patterns are configured.	Blocks all nonlocal Call Transfers.	Allows all nonlocal Call Transfers for backward compatibility.
Specific transfer patterns are configured.	Allows Call Transfers to specific external entities.	Allows Call Transfers to specific external entities.
The **transfer-pattern blocked** command is configured.	Blocks all nonlocal Call Transfers are blocked. **Note** The configuration reverts to the default, where no transfer patterns are configured.	All nonlocal Call Transfers are blocked. **Note** The configuration unconditionally blocks all nonlocal Call Transfers. It does not return to the default, where all nonlocal Call Transfers are allowed.

Conference-Pattern Blocked

The **conference-pattern blocked** command is used to prevent extensions on a voice register Pool from initiating conferences.

The following table summarizes the behavior of the **conference-pattern blocked** command in relation to **no conference-pattern blocked** ,**conference max-length** , **no conference max-length** , and **transfer max-length** commands.

	Conference max-length	No conference max-length
No conference-pattern blocked (default case)	Allowing/Blocking of conference call depends on configured conference max-length.	Allowing/Blocking of conference call depends on configured transfer max-length.
Conference-pattern blocked	Conference calls are not allowed on SIP and SCCP phones.	

	Max-length <= allowed max-length		Max-length > allowed max-length	
	Transfer	Conference	Transfer	Conference
Transfer max-length + No Conference max-length (use transfer max-length for conference cases too, as conference max-length not configured)	Y	Y	N	N

	Max-length <= allowed max-length		Max-length > allowed max-length	
No transfer max-length + Conference max-length (conference max-length has precedence over transfer max-length for conference)	Y	Y	Y	N
No transfer max-length + Conference max-length (conference max-length has precedence over transfer max-length for conference)	Y	Y	N	N
No transfer max-length + No conference max-length	All transfer and conference calls are allowed.			

Configuring the Maximum Number of Digits for a Conference Call

Before you begin

Cisco Unified SRST 10.5 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *pool-tag* OR**ephone***phone-tag*
4. **conference max-length** *value*
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router# enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register pool** *pool-tag* OR**ephone***phone-tag* **Example:** `Router(config)# voice register pool 25`	Enters voice register Pool configuration mode and creates a Pool configuration for a Cisco Unified SIP IP phone in Cisco Unified Communications Manager Express or for a set of Cisco Unified SIP IP phones in Cisco Unified SIP SRST.

	Command or Action	Purpose
		• *pool-tag* : Unique number assigned to the Pool. Range is 1–100. OR Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones. • *template-tag* : Declares a template tag. Range is 1–10. OR Enters ephone configuration mode. • *phone-tag* : Unique sequence number that identifies this ephone during configuration tasks. The maximum number of ephones is version and platform-specific. Type? To display range.
Step 4	**conference max-length** *value* **Example:** `Router(config-telephony)# conference max-lenght 6`	Allows the conference of calls from Cisco IP phones to specified directory numbers of phones other than Cisco IP phones. *conference max-length* Allows conference call depending on the configured conference max-length. Range is 3–16.
Step 5	**end** **Example:** `Router(config-telephony)# end`	Exits telephony-service configuration mode and enter privileged EXEC mode.

Configuring Conference Blocking Options for Phones

Before you begin

- Use Cisco Unified SRST 10.5 or a later version.
- Configure the transfer-pattern command.
- Configure the conference transfer-pattern command.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *pool-tag* OR**ephone***phone-tag*
4. **conference-pattern blocked**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router# enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register pool** *pool-tag* OR**ephone***phone-tag* **Example:** `Router(config)# voice register pool 25`	Enters voice register Pool configuration mode and creates a Pool configuration for a Cisco Unified SIP IP phone in Cisco Unified Communications Manager Express or for a set of Cisco Unified SIP IP phones in Cisco Unified SIP SRST. • *pool-tag* : Unique number assigned to the Pool. Range is 1–100. OR Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones. • *template-tag* : Declares a template tag. Range is 1–10. OR Enters ephone configuration mode. • *phone-tag* : Unique sequence number that identifies this ephone during configuration tasks. The maximum number of ephones is version and platform-specific. Type? To display range.
Step 4	**conference-pattern blocked** **Example:** `Router(config-telephony)# conference-pattern blocked`	Allows the conference of calls from Cisco IP phones to specified directory numbers of phones other than Cisco IP phones. *conference-pattern blocked* No conference calls are allowed. .
Step 5	**exit** **Example:** `Router(config-telephony)# exit`	Exits telephony-service configuration mode and enter global configuration mode.

Transfer-Pattern Blocked

When the **transfer-pattern blocked** command is configured for a specific phone, no Call Transfers are allowed from that phone over the trunk.

This feature forces unconditional blocking of all Call Transfers from the specific phone to any other nonlocal numbers (external calls from one trunk to another trunk). No Call Transfers from this specific phone are possible even when a transfer pattern matches the dialed digits for transfer.

The following table compares the behaviors of Cisco Unified SCCP and SIP IP phones for specific configurations.

Configuration	Cisco Unified SCCP IP Phones	Cisco Unified SIP IP Phones
No transfer patterns are configured.	Blocks all nonlocal Call Transfers.	Allows all nonlocal Call Transfers for backward compatibility.
Specific transfer patterns are configured.	Allows Call Transfers to specific external entities.	Allows Call Transfers to specific external entities.
The **transfer-pattern blocked** command is configured.	Blocks all nonlocal Call Transfers are blocked. **Note** The configuration reverts to the default, where no transfer patterns are configured.	All nonlocal Call Transfers are blocked. **Note** The configuration unconditionally blocks all nonlocal Call Transfers. It does not return to the default, where all nonlocal Call Transfers are allowed.

Conference Transfer-Pattern

When both the **transfer-pattern** and **conference transfer-pattern** commands are configured and dialed digits match the configured transfer pattern, conference calls are allowed. However, when the dialed digits do not match the configured transfer pattern, the conference call is blocked.

For information on provisioning Cisco Unified IP phones for secure access to web content using HTTPS, see the HTTPS Provisioning for Cisco Unified IP Phones section of Cisco Unified Communications Manager Express System Administrator Guide.

For configuration examples, see the Configuring HTTPS Support for Cisco Unified Communications Manager Express: Example section of Cisco Unified Communications Manager Administration Guide.

New Features in Cisco Unified SRST Version 9.1

Cisco Unified SRST 9.1 supports the following new features:

- Key Expansion Module Support for Cisco Unified SIP IP Phones
- Enhancement in Speed-Dial Support
- Voice Hunt Group Support

> **Note** If you have older routers, such as the VG26nn and VG37nn platforms and Cisco Integrated Services Router (ISR) Generation 1 platforms (Cisco ISR 1861, 2800, and 3800 Series), you must upgrade to Cisco ISR 881, 886VA, 887VA, 888, 888E, 1861E, 2900, 3900, and 3900E Series platforms to utilize these new features.

Key Expansion Module Support for Cisco Unified SIP IP Phones

Cisco Unified IP Key Expansion Modules (KEMs) are supported on Cisco Unified 8851/51NR, 8861, 8961, 9951, and 9971 SIP IP phones from Cisco Unified SIP SRST 9.1.

For information on KEMs support for Cisco Unified 8851/51NR, 8861, 8961, 9951, and 971 SIP IP phones, see Phone Feature Support Guide for Cisco Unified Communications Manager Express, Cisco Unified SRST, Unified E-SRST, and Unified Secure SRST.

Restrictions

- Bulk registration is not supported for KEMs in Cisco Unified SRST. Phones do not send bulk Registration Requests but always use the UDP port for registration.
- KEMs is not supported for Cisco Unified SCCP IP Phones and Cisco Unified SIP IP Phones other than the Cisco Unified 8851/51NR, 8861, 8961, 9951, and 9971 SIP IP phones.
- Features configured on keys are disabled when supported Cisco Unified SIP IP phones are in Cisco Unified SIP SRST.
- All Cisco Unified 8851/51NR, 8861,8961, 9951, and 9971 SIP IP phone restrictions and limitations apply to KEMs.
- All Cisco Unified SIP SRST feature restrictions and limitations apply to KEMs.

For more information on how the **blf-speed-dial** , **number** , and **speed-dial** commands, in voice register Pool configuration mode, have been modified, see Cisco Unified Communications Manager Express Command Reference.

For information on installing KEMs on Cisco Unified IP Phone, see the Installing a Key Expansion Module on the Cisco Unified IP Phone section of Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 7.1 (3) (SIP).

For information on installing KEMs on Cisco Unified 8811, 8841, 8851, 8851NR, and 8861 Phones, see the Cisco IP Phone Key Expansion Module section of Cisco IP Phone 8811, 8841, 8851, 8851NR, and 8861 Administration Guide for Cisco Unified Communications Manager.

Enhancement in Speed-Dial Support

Cisco Unified SRST 9.1 ignores the “,” or comma (pause indicator) to avoid break-in speed-dial support.

Because the pause speed-dial feature (supported in Cisco Unified Communications Manager or Cisco Unified Communications Manager) is not supported in Cisco Unified SRST, Cisco Unified Communications Manager and phones (Cisco Unified SCCP IP phones and Cisco Unified SIP IP phones) registered in Cisco Unified SRST maintain backward compatibility in Cisco Unified SRST mode. When phones failover to the Cisco Unified SRST router during WAN outages and Cisco Unified Communications Manager fails, the phones only send the speed-dial numbers when the pause speed-dial buttons are pressed. The comma pause indicator is ignored and the preconfigured FAC, PIN, and DTMF are not sent.

For information on configuring speed-dial in Cisco Unified Communications Manager, see the "Device setup" chapter of Cisco Unified Communications Manager Administration Guide.

Voice Hunt Group Support

Cisco Unified SIP SRST 9.1 supports voice hunt groups. Voice hunt groups allow call placed to a single (pilot) number to contact multiple destinations.

There are three different types of voice hunt groups. Each type uses a different strategy to determine the first number that rings for successive calls to the pilot number until a number answers.

- Parallel Hunt Groups—Allows an incoming call to simultaneously ring all the numbers in the hunt group member list.
- Sequential Hunt Groups—Allows an incoming call to ring all the numbers in the left-to-right order in which they were listed while defining the hunt group. The first number in the list is always the first number tried when the pilot number is called. Maximum number of hops is not a configurable parameter for sequential hunt groups.
- Longest-idle Hunt Groups—Allows an incoming call to first go to the number that has been idle the longest for the number of hops specified when the hunt group was defined. The longest-idle time is determined from the last time that a phone registered, reregistered, or went on-hook.

Cisco Unified SCCP IP phones support only ephone hunt groups whereas a voice hunt group supports Cisco Unified SCCP IP phones, Cisco Unified SIP IP phones. In addition, it also supports a mixture of Cisco Unified SCCP IP phones and Cisco Unified SIP IP phones.

With the voice hunt group feature preconfigured in the Cisco Unified SIP SRST router, voice hunt groups continue to be supported after phones fallback from Cisco Unified Communications Manager to the Cisco Unified SIP SRST router.

Restrictions

- Hunt group statistics is not supported for voice hunt groups in Cisco Unified SRST.
- Hunt group nesting or setting the final number of one hunt groups as the pilot of another hunt group is not supported.

New Features in Cisco Unified SRST Version 9.0

Support for Cisco Unified 6901 and 6911 SIP IP Phones

For information on feature support for the Cisco Unified 6901 and 6911SIP IP Phones in Cisco Unified SRST, see Phone Feature Support Guide for Cisco Unified Communications Manager Express, Cisco Unified SRST, Unified E-SRST, and Unified Secure SRST.

Support for Cisco Unified 6921, 6941, 6945, and 6961 SIP IP Phones

For information on feature support for the Cisco Unified 6921, 6941, 6945, and 6961 SIP IP Phones in Cisco Unified SRST, see Phone Feature Support Guide for Cisco Unified Communications Manager Express, Cisco Unified SRST, Unified E-SRST, and Unified Secure SRST.

Support for Cisco Unified 8941 and 8945 SIP IP Phones

For information on feature support for the Cisco Unified 8941 and 8945 SIP IP Phones in Cisco Unified SRST, see Phone Feature Support Guide for Cisco Unified Communications Manager Express, Cisco Unified SRST, Unified E-SRST, and Unified Secure SRST.

Multiple Calls Per Line

Cisco Unified SRST 9.0 supports the Multiple Calls Per Line (MCPL) feature on Cisco Unified 6921, 6941, 6945, and 6961 SIP IP phones. In addition, it supports Cisco Unified 8941, 8945 SCCP, and SIP IP phones.

Before Cisco Unified SRST 9.0, supports only two calls for every directory number (DN) on Cisco Unified 8941 and 8945 SCCP IP phones.

With Cisco Unified SRST 9.0, the MCPL feature overcomes the limitation on the maximum number of calls per line.

Cisco Unified SRST 9.0 does not support the MCPL feature on Cisco Unified 6921, 6941, 6945, and 6961 SCCP IP phones. Allows only two calls on these phones whereas allows only one call on octo-line directory numbers on these phones before activating Call Forward Busy or busy tone.

Cisco Unified 8941 and 8945 SCCP IP Phones

Before Cisco Unified SRST 9.0, the values for the **max-dn** and **timeouts busy** commands were hardcoded for Cisco Unified 8941 and 8945 SCCP IP phones.

In Cisco Unified SRST 9.0, you can configure the **max-dn** and**timeouts busy** commands in call-manager-fallback configuration mode. Use the **max-dn** command to set the maximum number of DNs that can be supported by the router and enable dual-line mode, octo-line mode, or both modes. Use the **timeouts busy** command to set the timeout value for Call Transfers to busy destinations.

For configuration information, see the “Configuring the Maximum Number of Calls” section on page 162.

Cisco Unified 6921, 6941, 6945, 6961, 8941, and 8945 SIP IP Phones

In Cisco Unified SRST 9.0, the maximum number of calls for Cisco Unified 6921, 6941, 6945, 6961, 8941, and 8945 SIP IP phones is controlled by the phones.

Prerequisites

- Cisco Unified SRST 9.0 and later versions.
- Correct firmware is installed:
  - 9.2(1) or a later version for Cisco Unified 6921, 6941, 6945 and 6961 SIP IP phones.
  - 9.2(2) or a later version for Cisco Unified 8941 and 8945 SIP IP phones.

Voice and Fax Support on Cisco ATA-187

Cisco ATA-187 is a SIP-based analog phone adapter that turns traditional phone devices into IP devices. Cisco ATA-187 can connect with a regular analog FXS phone or fax machine on one end, while the other end is an IP side that uses SIP for signaling and registers as a Cisco Unified SIP IP phone.

Cisco ATA-187 functions as a Cisco Unified SIP IP phone that supports T.38 fax relay and fax pass-through, enabling the real-time transmission of fax over IP networks. The fax rate is from 7.2 to 14.4 kbps.

For information on feature support for the Cisco ATA-187 in Cisco Unified SRST, see Phone Feature Support Guide for Cisco Unified Communications Manager Express, Cisco Unified SRST, Unified E-SRST, and Unified Secure SRST.

For more information on Cisco ATA-187, see Cisco ATA 187 Analog Telephone Adaptor Administration Guide for SIP.

New Features in Cisco Unified SRST Version 8.8

Cisco Unified SRST 8.8 supports the following new Cisco Unified SCCP IP phones:

- Cisco Unified 6945 SCCP IP Phones
- Cisco Unified 8941 SCCP IP Phones
- Cisco Unified 8945 SCCP IP Phones

Support for Cisco Unified 6945, 8941, and 8945 SCCP IP Phones

For information on feature support for the Cisco Unified 6945, 8941, and 8945 SCCP IP Phones in Cisco Unified SRST, see Phone Feature Support Guide for Cisco Unified Communications Manager Express, Cisco Unified SRST, Unified E-SRST, and Unified Secure SRST.

For information on the Cisco Unified 6945 SCCP IP Phone, see Cisco Unified IP Phone 6945 User Guide for Cisco Unified Communications Manager Express Version 8.8 (SCCP).

For information on the Cisco Unified 8941 and 8945 SCCP IP Phones, see Cisco Unified IP Phone 8941 and 8945 User Guide for Cisco Unified Communications Manager Express Version 8.8 (SCCP).

New Features in Cisco Unified SRST Version 8.0

Beginning with Cisco IP Phone firmware 8.5(3) and Cisco IOS Release 15.1(1)T, Cisco Unified SRST supports SIP signaling over UDP, TCP, and TLS connections, providing both RTP and SRTP media connections based on the security settings of the IP phone.

New Features in Cisco Unified SRST Version 7.0/4.3

Cisco Unified SRST 7.0/4.3 supports the following new features:

- Configuring Eight Calls per Button (Octo-Line)

• Configuring Consultative Transfer

New Features in Cisco Unified SRST Version 4.2(1)

Cisco Unified SRST Version 4.2(1) introduces the new feature enhancements for Enhanced 911 Services.

New Features in Cisco Unified SRST Version 4.1

Cisco Unified SRST Version 4.1 introduces the following new feature:

• Enhanced 911 Services

New Features in Cisco Unified SRST Version 4.0

Additional Cisco Unified IP Phone Support

The following IP phones are supported with Cisco Unified SRST systems:

• Cisco Unified IP Phone 7911G

• Cisco Unified IP Phone 7941G and Cisco Unified IP Phone 7941G-GE

• Cisco Unified IP Phone 7960G

• Cisco Unified IP Phone 7961G and Cisco Unified IP Phone 7961G-GE

In addition, the Cisco Unified IP Phone 7914 Expansion Module can attach to the Cisco 7941G-GE and Cisco 7961G-GE. The Cisco 7914 Expansion Module adds additional features, such as adding 14 line appearances or speed-dial numbers to your phone. You can attach one or two expansion modules to your IP phone. When you use two expansion modules, you have 28 additional line appearances or speed-dial numbers, or a total of 34 line appearances or speed-dial numbers. For more information, see Cisco IP Phone 7914 Expansion Module Quick Start Guide.

No additional SRST configuration is required for these phones.

The **show ephone** command is enhanced to display the configuration and status of the new Cisco IP Phones added to SRST Version 4.0. For more information, see the **show ephone** command in Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

To determine compatible firmware, platforms, memory, and additional voice products that are associated with Cisco Unified SRST 4.0, see Cisco Unified SRST 4.3 Supported Firmware, Platforms, Memory, and Voice Products.

Cisco IP Communicator Support

Cisco IP Communicator is a software-based application that delivers enhanced telephony support on personal computers. This SCCP-based application allows computers to function as IP phones, providing high-quality voice calls on the road, in the office, or from wherever users may have access to the corporate network. Cisco

IP Communicator appears on a user's computer monitor as a graphical, display-based IP phone with a color screen, a key pad, feature buttons, and soft keys.

Fax Pass-through using SCCP and ATAs Support

Fax pass-through mode is now supported using Cisco VG 224 voice gateways, Analog Telephone Adaptors (ATA), and SCCP. ATAs ship with SIP firmware, so SCCP firmware must be loaded before this feature can be used.

Note For ATAs that are registered to a Cisco Unified SRST system to participate in FAX calls, they must have their ConnectMode parameter set to use the "standard payload type 0/8" as the RTP payload type in FAX pass-through mode. For ATAs used with Cisco Unified SRST 4.0 and higher versions, this is done by setting bit 2 of the ConnectMode parameter to 1 on the ATA. For more information, see the "Parameters and Defaults" chapter in Cisco ATA 186 and Cisco ATA 188 Analog Telephone Adaptor Administrator's Guide for SCCP.

H.323 VoIP Call Preservation Enhancements for WAN Link Failures for SCCP Phones

H.323 VoIP call preservation enhancements for WAN link failures sustains connectivity for H.323 topologies where signaling is handled by an entity, such as Cisco Unified Communications Manager, that is different from the other endpoint and brokers signaling between the two connected parties.

Call preservation is useful when a gateway and the other endpoint (typically a Cisco Unified IP phone) are collocated at the same site and the call agent is remote and therefore more likely to experience connectivity failures. H.323 VoIP call preservation enhancements does not support SIP Phones.

For configuration information see the "Configuring H.323 Gateways" chapter in Cisco IOS H.323 Configuration Guide.

Video Support

This feature allows you to set video parameters for the Cisco Unified SRST to maintain close feature parity with Cisco Unified CM. When the Cisco Unified SRST is enabled, Cisco Unified IP Phones do not have to be reconfigured for video capabilities because all ephones retain the same configuration used with Cisco Unified CM. However, you must enter call-manager-fallback configuration mode to set video parameters for Cisco Unified SRST. The feature set for video is the same as that for Cisco Unified SRST audio calls.

For more information, see the "Setting Video Parameters" section on page 357.

New Features in Cisco Unified SRST Version 3.4

Cisco SIP SRST 3.4

Cisco SIP SRST Version 3.4 describes SRST functionality for Session Initiation Protocol (SIP) networks. Cisco SIP SRST Version 3.4 provides backup to an external SIP call control (IP-PBX) by providing basic registrar and back-to-back user agent (B2BUA) services. These services are used by a SIP IP phone in the event of a WAN connection outage when the SIP phone is unable to communicate with its primary SIP proxy.

Cisco SIP SRST Version 3.4 can support SIP phones with standard RFC 3261 feature support locally and across SIP WAN networks. With Cisco SIP SRST Version 3.4, SIP phones can place calls across SIP networks in the same way as Skinny Client Control Protocol (SCCP) phones. For full information about SIP SRST, Version 3.4, see Cisco SIP SRST Version 3.4 System Administrator Guide.

New Features in Cisco SRST Version 3.3

Secure SRST

Secure Cisco IP phones that are located at remote sites and that are attached to gateway routers can communicate securely with Cisco Unified Communications Manager using the WAN. But if the WAN link or Cisco Unified Communications Manager goes down, all communication through the remote phones becomes nonsecure. To overcome this situation, gateway routers can now function in secure SRST mode, which activates when the WAN link or Cisco Unified Communications Manager goes down. When the WAN link or Cisco Unified Communications Manager is restored, Cisco Unified Communications Manager resumes secure call-handling capabilities.

Secure SRST provides new SRST security features such as authentication, integrity, and media encryption. Authentication provides assurance to one party that another party is whom it claims to be. Integrity provides assurance that the given data has not been altered between the entities. Encryption implies confidentiality; that is, that no one can read the data except the intended recipient. These security features allow privacy for SRST voice calls and protect against voice security violations and identity theft. For more information see the “Configuring Secure SRST for SCCP and SIP” section on page 241.

Cisco Unified IP Phone 7970G and Cisco Unified 7971G-GE Support

The Cisco Unified IP Phones 7970G and 7971G-GE are full-featured telephones that provide voice communication over an IP network. They function much like a traditional analog telephones, allowing you to place and receive phone calls and to access features such as mute, hold, transfer, speed dial, call forward, and more. In addition, because the phones are connected to your data network, they offer enhanced IP telephony features, including access to network information and services, and customizable features and services. The phones also support security features that include file authentication, device authentication, signaling encryption, and media encryption.

The Cisco Unified IP Phones 7970G and 7971G-GE also provide a color touchscreen, support for up to eight line or speed-dial numbers, context-sensitive online help for buttons and feature, and a variety of other sophisticated functions. No configurations specific to SRST are necessary.

For more information, see the Cisco Unified IP Phone 7900 Series documentation index.

Note The Cisco Unified IP Phone 7914 Expansion Module can attach to your Cisco Unified IP Phones 7970G and 7971G-GE. See the “Cisco Unified IP Phone Expansion Module 7914 Support” section on page xliv for more information.

Enhancement to the show ephone Command

The **show ephone** command is enhanced to display the configuration and status of the Cisco Unified IP Phone 7970G and Cisco Unified IP Phone 7971G-GE. For more information, see the **show ephone** command in Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

New Features in Cisco SRST Version 3.2

Enhancement to the alias Command

The **alias** command is enhanced as follows:

- The **cfw** keyword was added, providing call forward no-answer/busy capabilities.
- The maximum number of **alias** commands used for creating calls to telephone numbers that are unavailable during Cisco Unified Communications Manager fallback was increased to 50.
- The *alternate-number* argument can be used in multiple **alias** commands.

For more information, see the **alias** command in Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

Enhancement to the cor Command

The maximum number of **cor** lists has increased to 20.

For more information, see the **cor** command in Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

Enhancement to the pickup Command

The **pickup** command was introduced to enable the PickUp soft key on all Cisco Unified IP Phones, allowing an external Direct Inward Dialing (DID) call coming into one extension to be picked up from another extension during SRST.

For more information, see the **pickup** command in Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

Enhancement to the user-locale Command

The **user-locale** command is enhanced to display the Japanese Katakana country code. Japanese Katakana is available in Cisco Unified Communications Manager V4.0 or later versions.

For more information, see the **user-locale** command in the Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

Increased the Number of Cisco Unified IP Phones Supported on the Cisco 3845

The Cisco 3845 now supports 720 phones and up to 960 ephone-dns or virtual voice ports.

MOH Live-Feed Support

Cisco Unified SRST is enhanced with the new **moh-live** command. The **moh-live** command provides live-feed MOH streams from an audio device connected to an E&M or FXO port to Cisco IP phones in SRST mode. If an FXO port is used for a live feed, the port must be supplied with an external third-party adaptor to provide a battery feed. Music from a live feed is obtained from a fixed source and is continuously fed into the MOH playout buffer instead of being read from a flash file. Live-feed MOH can also be multicast to Cisco IP phones. See the “Integrating Cisco Unified Communications Manager and Cisco Unified SRST to Use Cisco Unified SRST as a Multicast MOH Resource” section on page 11 for configuration instructions.

No Timeout for Call Preservation

To preserve existing H.323 calls on the branch in the event of an outage, disable the H.225 keepalive timer by entering the **no h225 timeout keepalive** command. This feature is supported in Cisco IOS Releases 12.3(7)T1 and higher versions. See the “Cisco Unified SRST Feature Overview” section on page 1 for more information.

H.323 is not supported with SIP phones.

RFC 2833 DTMF Relay Support

Cisco Skinny Client Control Protocol (SCCP) phones, such as those used with Cisco 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 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. See the “How to Configure DTMF Relay for SIP Applications and Voicemail” section on page 352 for configuration instructions.

To use voicemail 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.

Translation Profile Support

Cisco SRST 3.2 and later versions support translation profiles. Translation profiles allow you to group translation rules together and to associate translation rules with the following:

- Called numbers
- Calling numbers
- Redirected called numbers

See the “Enabling Translation Profiles” section on page 202 for more configuration information. For more information on the **translation-profile** command, see Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

New Features in Cisco Unified SRST Version 3.1

Cisco Unified SRST V3.1 introduced the new features described in the following sections:

- Cisco Unified IP Phone 7920 Support
- Cisco Unified IP Phone 7936 Support

Note For information about Cisco Unified IP phones, see the Cisco Unified IP Phone 7900 Series documentation.

Cisco Unified IP Phone 7920 Support

The Cisco Unified Wireless IP Phone 7920 is an easy-to-use IEEE 802.11b wireless IP phone that provides comprehensive voice communications in conjunction with Cisco Unified CM and Cisco Aironet 1200, 1100, 350, and 340 Series of Wi-Fi (IEEE 802.11b) access points. As a key part of the Cisco AVVID Wireless Solution, the Cisco Unified Wireless IP Phone 7920 delivers seamless intelligent services, such as security, mobility, quality of service (QoS), and management, across an end-to-end Cisco network.

No configuration is necessary.

Cisco Unified IP Phone 7936 Support

The Cisco Unified IP Conference Station 7936 is an IP-based, hands-free conference room station that uses VoIP technology. The IP Conference Station replaces a traditional analog conferencing unit by providing business conferencing features—such as call hold, call resume, call transfer, call release, redial, mute, and conference—over an IP network.

No configuration is necessary.

New Features in Cisco SRST Version 3.0

Additional Language Options for IP Phone Display

Displays for the Cisco Unified IP Phone 7940G and Cisco Unified IP Phone 7960G can be configured with extra ISO-3166 codes for German, Danish, Spanish, French, Italian, Japanese, Dutch, Norwegian, Portuguese, Russian, Swedish, United States.

Note This feature is available only for Cisco Unified SRST running under Cisco Unified Communications Manager V3.2.

Consultative Call Transfer and Forward Using H.450.2 and H.450.3 for SCCP Phones

Cisco Unified SRST V1.0, Cisco Unified SRST V2.0, and Cisco Unified SRST V2.1 allow blind Call Transfers and blind call forwarding. Blind calls do not give transferring and forwarding parties the ability to announce or consult with destination parties. These three versions of Cisco Unified SRST use a Cisco Unified SRST proprietary mechanism to perform blind transfers. Cisco Unified SRST V3.0 adds the ability to perform Call Transfers with consultation using the ITU-T H.450.2 (H.450.2) standard and call forwarding using the ITU-T H.450.3 (H.450.3) standard for H.323 calls.

Cisco Unified SRST V3.0 provides support for IP phones to initiate Call Transfer and forwarding with H.450.2 and H.450.3 by using the default session application. The built-in H.450.2 and H.450.3 support that is provided by the default session application applies to Call Transfers and call forwarding initiated by IP phones, regardless of the PSTN interface type.

Note All voice gateway routers in the VoIP network must support H.450. For H.450 support, routers with Cisco Unified SRST must run either Cisco Unified SRST V3.0 and higher versions or Cisco IOS Release 12.2(15)ZJ and later releases. Routers without Cisco Unified SRST must run either Cisco Unified SRST V2.1 and higher versions or Cisco IOS Release 12.2(11)YT and later releases. SIP phones do not support this feature.

For more information about the default session application, see the Default Session Application Enhancements Guide.

For configuration information, see the “Enabling Consultative Call Transfer and Forward Using H.450.2 and H.450.3 with Cisco Unified SRST 3.0” section on page 210.

Customized System Message for Cisco Unified IP Phones

The display message that appears on Cisco Unified IP Phone 7905G, Cisco Unified IP Phone 7940G, Cisco Unified IP Phone 7960G, and Cisco Unified IP Phone 7910 units when they are in fallback mode can be customized. The new system message command allows you to edit these display messages on a per-router basis. The custom system message feature supports English only.

For further information, see the “Configuring Customized System Messages for Cisco Unified IP Phones” section on page 156.

Dual-Line Mode

A new keyword that was added to the **max-dn** command allows you to set IP phones to dual-line mode. Each dual-line IP phone must have one voice port and two channels to handle two independent calls. This mode enables call waiting, Call Transfer, and conference functions on a single ephone-dn (ephone directory number). There is a maximum number of DNs available during Cisco Unified SRST fallback. The **max-dn** command affects all IP phones on a Cisco Unified SRST router.

For configuration information, see the “Configuring Dual-Line Phones” section on page 158.

E1 R2 Signaling Support

Cisco Unified SRST V3.0 supports E1 R2 signaling. R2 signaling is an international signaling standard that is common to channelized E1 networks; however, there is no single signaling standard for R2. The ITU-T Q.400-Q.490 recommendation defines R2, but several countries and geographic regions implement R2 in entirely different ways. Cisco addresses this challenge by supporting many localized implementations of R2 signaling in its Cisco IOS Software.

The Cisco E1 R2 signaling default is ITU, which supports the following countries: Denmark, Finland, Germany, Russia (ITU variant), Hong Kong (ITU variant), and South Africa (ITU variant). The expression "ITU variant" means that there are multiple R2 signaling types in the specified country, but Cisco supports the ITU variant.

Cisco also supports specific local variants of E1 R2 signaling in the following regions, countries, and corporations:

- Argentina
- Australia
- Bolivia
- Brazil
- Bulgaria
- China
- Colombia
- Costa Rica
- East Europe (includes Croatia, Russia, and Slovak Republic)
- Ecuador (ITU)
- Ecuador (LME)
- Greece
- Guatemala
- Hong Kong (uses the China variant)
- Indonesia
- Israel
- Korea
- Laos
- Malaysia
- Malta
- New Zealand
- Paraguay
- Peru

- Philippines
- Saudi Arabia
- Singapore
- South Africa (Panaftel variant)
- Telmex Corporation (Mexico)
- Telnor Corporation (Mexico)
- Thailand
- Uruguay
- Venezuela
- Vietnam

European Date Formats

The date format on a Cisco IP phone display can be configured with the following two extra formats:

- yy-mm-dd (year-month-day)
- yy-dd-mm (year-day-month)

For configuration information, see the “Configuring IP Phone Clock, Date, and Time Formats” section on page 152.

Huntstop for Dual-Line Mode

A new keyword was added to the huntstop command. The **channel** keyword causes hunting to skip the secondary channel in dual-line configuration if the primary line is busy or does not answer.

For configuration information, see the “Configuring Dial-Peer and Channel Hunting” section on page 206.

Music On Hold for Multicast from Flash Files

You can configure Cisco Unified SRST to support continuous multicast output of MOH from a flash MOH file in flash memory.

For more information, see the “Defining XML API Schema” section on page 238.

Ringing Timeout Default

A ringing timeout default can be configured for extensions on which no-answer call forwarding has not been enabled. Expiration of the timeout causes incoming calls to return a disconnect code to the caller. This mechanism provides protection against hung calls for inbound calls received over interfaces such as Foreign Exchange Office (FXO) that do not have forward-disconnect supervision. For more information, see the “Configuring the Ringing Timeout Default” section on page 208.

Secondary Dial Tone

Secondary dial tone is available for Cisco Unified IP Phones running Cisco Unified SRST. The secondary dial tone is generated when you dial a predefined PSTN access prefix. For example, you would hear different dial tone when a designated number is pressed to reach an outside line.

The secondary dial tone is created through the secondary dial tone command. For more information, see the “Configuring a Secondary Dial Tone” section on page 157.

Enhancement to the Show ephone Command

The **show ephone** command is enhanced to display the following:

- Configuration and status of additional phones (new keywords: **7905, 7914, 7935, ATA**)
- Status of all phones with the call-forwarding all (CFA) feature enabled on at least one of their DNs (new keyword: **cfa**)

For more information, see the **show ephone** command in Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

System Log Messages for Phone Registrations

Diagnostic messages are added to the system log whenever a phone registers or unregisters from Cisco Unified SRST.

Three-Party G.711 Ad Hoc Conferencing

Cisco Unified SRST supports three-party instant meeting conferencing using the G.711 coding technique. For conferencing to be available, connect two lines to one or more buttons of an IP phone.

For more information, see the “Enabling Three-Party G.711 Ad Hoc Conferencing” section on page 236.

Support for Cisco VG248 Analog Phone Gateway 1.2(1) and Higher Versions

The Cisco VG248 Analog Phone Gateway is a mixed-environment solution, enabled by Cisco Unified Communications system. It allows organizations to support their legacy analog devices while taking advantage of the new opportunities afforded by using IP telephony. The Cisco VG248 is a high-density gateway for using analog phones, fax machines, modems, voicemail systems, and speakerphones within an enterprise voice system based on Cisco Unified Communications Manager.

During Cisco Unified Communications Manager fallback, Cisco Unified SRST considers the Cisco VG248 to be a group of Cisco Unified IP Phones. Cisco Unified SRST counts each of the 48 ports on the Cisco VG248 as a separate Cisco Unified IP Phone. Support for Cisco VG248 Version 1.2(1) and higher versions is also available in Cisco Unified SRST Version 2.1.

For more information, see Cisco VG248 Analog Phone Gateway Data Sheet and Cisco VG248 Analog Phone Gateway Version 1.2(1) Release Notes.

New Features in Cisco SRST Version 2.1

Cisco SRST V2.1 introduced the new features described in the following sections:

- Additional Language Options for IP Phone Display
- Cisco Unified SRST Aggregation
- Cisco ATA 186 and ATA 188 Support
- Cisco Unified IP Phone 7902G Support
- Cisco Unified IP Phone 7905G Support
- Cisco Unified IP Phone 7912G Support
- Cisco Unified IP Phone Expansion Module 7914 Support
- Enhancement to the Dial Plan-Pattern Command

Note For information about Cisco Unified IP phones, see the Cisco Unified IP Phone 7900 Series documentation.

Additional Language Options for IP Phone Display

Displays for the Cisco Unified IP Phone 7940G and Cisco Unified IP Phone 7960G can be configured with ISO-3166 codes for the following countries:

- France
- Germany
- Italy
- Portugal
- Spain
- United States

Note This feature is available only in Cisco Unified SRST running under Cisco Unified Communications Manager V3.2.

For configuration information, see the "Configuring IP Phone Language Display" section on page 154.

Cisco Unified SRST Aggregation

For systems running Cisco Unified Communications Manager 3.3(2) and later versions, the restriction of running Cisco Unified SRST on a default gateway was removed. Multiple SRST routers can be used to support more phones. Carefully plan and configure the dial peers and dial plans for Call Transfer and forwarding to work properly.

Cisco ATA 186 and ATA 188 Support

The Cisco ATA analog phone adapters are handset-to-Ethernet adapters that allow regular analog phones to operate on IP-based telephony networks. Cisco ATAs support two voice ports, each with an independent phone number. The Cisco ATA 188 also has an RJ-45 10/100BASE-T data port. Cisco Unified SRST supports Cisco ATA 186 and Cisco ATA 188 using Skinny Client Control Protocol (SCCP) for the voice calls only.

Cisco Unified IP Phone 7902G Support

The Cisco Unified IP Phone 7902G is an entry-level IP phone that addresses the voice communications needs of a lobby, laboratory, manufacturing floor, hallway, or other area where only basic calling capability is required.

The Cisco Unified IP Phone 7902G is a single-line IP phone with fixed feature keys that provide one-touch access to the redial, transfer, conference, and voicemail access features. Consistent with other Cisco IP phones, the Cisco Unified IP Phone 7902G supports inline power, which allows the phone to receive power over the LAN. This capability gives the network administrator centralized power control and thus greater network availability.

Cisco Unified IP Phone 7905G Support

The Cisco Unified IP Phone 7905G is a basic IP phone that provides a core set of business features. It provides single-line access and four interactive softkeys that guide a user through call features and functions via the pixel-based LCD. The graphic capability of the display presents calling information, intuitive access to features, and language localization in future firmware releases. The Cisco Unified IP Phone 7905G supports inline power, which allows the phone to receive power over the LAN.

No configuration is necessary.

Cisco Unified IP Phone 7912G Support

The Cisco Unified IP Phone 7912G provides core business features and addresses the communication needs of a cubicle worker who conducts low to medium phone traffic. Four dynamic softkeys provide access to call features and functions. The graphic display shows calling information and allows access to features.

The Cisco Unified IP Phone 7912G supports an integrated Ethernet switch, providing LAN connectivity to a local PC. In addition, the Cisco Unified IP Phone 7912G supports inline power, which allows the phone to receive power over the LAN. This capability gives the network administrator centralized power control and thus greater network availability. The combination of inline power and Ethernet switch support reduces cabling needs from a single wire to the desktop.

Cisco Unified IP Phone Expansion Module 7914 Support

The Cisco Unified IP Phone 7914 Expansion Module attaches to your Cisco Unified IP Phone 7960G, adding 14 line appearances or speed-dial numbers to your phone. You can attach one or two expansion modules to your IP phone. When you use two expansion modules, you have 28 additional line appearances or speed-dial numbers or a total of 34 line appearances or speed-dial numbers.

Enhancement to the Dial Plan-Pattern Command

A new keyword was added to the **dialplan-pattern** command. The extension-pattern keyword sets an extension number's leading digit pattern when it is different from the E.164 phone number's leading digits defined in the *pattern* variable. This enhancement allows manipulation of IP phone abbreviated extension number prefix digits. See the **dialplan-pattern** command in Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

New Features in Cisco SRST Version 2.02

Cisco Unified IP Phone Conference Station 7935 Support

The Cisco IP Conference Station 7935 is an IP-based, full-duplex hands-free conference station for use on desktops and offices and in small-to-medium-sized conference rooms. This device attaches a Cisco Catalyst 10/100 Ethernet switch port with a simple RJ-45 connection and dynamically configures itself to the IP network via the DHCP. Other than connecting the Cisco 7935 to an Ethernet switch port, no further administration is necessary. The Cisco 7935 dynamically registers to Cisco Unified CM for connection services and receives the appropriate endpoint phone number and any software enhancements or personalized settings, which are preloaded within Cisco Unified CM.

The Cisco Unified IP Phone 7935 provides three soft keys and menu navigation keys that guide a user through call features and functions. The Cisco Unified IP Phone 7935 also features a pixel-based LCD display. The display provides features such as date and time, calling party name, calling party number, digits dialed, and feature and line status. No configuration is necessary.

Increase in Directory Numbers

The following table shows the increases in directory numbers.

Cisco Router	Maximum Phones	Increase in Maximum Directory Number	
		From	To
Cisco 1751	24	96	120
Cisco 1760	24	96	120
Cisco 2600XM	24	96	120
Cisco 2691	72	216	288
Cisco 3640	72	216	288
Cisco 3660	240	720	960
Cisco 3725	144	432	576
Cisco 3745	240	720	960

Cisco Unity Voicemail Integration Using In-Band DTMF Signaling Across the PSTN and BRI/PRI

Cisco Unity voicemail and other voicemail systems can be integrated with Cisco Unified SRST. Voicemail integration introduces six new commands:

- Pattern direct
- Pattern ext-to-ext busy
- Pattern ext-to-ext no-answer
- Pattern trunk-to-ext busy
- Pattern trunk-to-ext no-answer
- Vm-integration