



Cisco Unified Survivable Remote Site Telephony Version 4.0

As the enterprise extends its IP telephony deployments from central sites to remote offices, one of the critical factors in achieving a successful deployment is the ability to support backup call control at the remote branch office. Cisco® Unified Survivable Remote Site Telephony (SRST) provides a cost-effective solution for supporting redundant call control in the remote branch office.

The Cisco Unified Communications system of voice and IP communications products and applications enables organizations to communicate more effectively-helping them to streamline business processes, reach the right resource the first time, and impact the top and bottom line. The Cisco Unified Communications portfolio is a key part of the Cisco Business Communications Solution-an integrated solution for organizations of all sizes which also includes network infrastructure, security, and network management products, wireless connectivity, and a lifecycle services approach, along with flexible deployment and outsourced management options, end-user and partner financing packages, and third-party communications applications.

BENEFITS OF CENTRALIZED CALL PROCESSING ARCHITECTURE

The Cisco Unified Survivable Remote Site Telephony product is a critical component of a centralized call-processing architecture in which a Cisco Unified CallManager cluster, located at a central site, provides telephony services for all sites of a corporation. The architecture provides numerous benefits to enterprises, including centralized and simplified management.

Table 1. Benefits of Centralized Call Processing Architecture

Centralized Call Processing Features	Benefits
Delivery of Full Feature Set to Remote Branches, Next-Generation Call Centers, Unified-Messaging Services, Embedded Directory Services, and Mobility	Improved productivity
Centralized Configuration and Management	Reduced operating expenses
Simplified Maintenance, and Troubleshooting	Reduced operating expenses
Converged Voice and Data Network	Reduced operating expenses
Reduced Installation Cost (Shared Cisco Unified CallManager Resource)	Reduced initial expense

However, centralized call processing architecture must include a strategy for survivability of telephony service at the branch office when access to the centralized call processing is interrupted due to WAN outage or other factors. Call-processing redundancy in the branch office is particularly critical in an emergency, which may be the cause of a WAN outage in the first place.

COMPONENTS OF CENTRALIZED CALL PROCESSING ARCHITECTURE

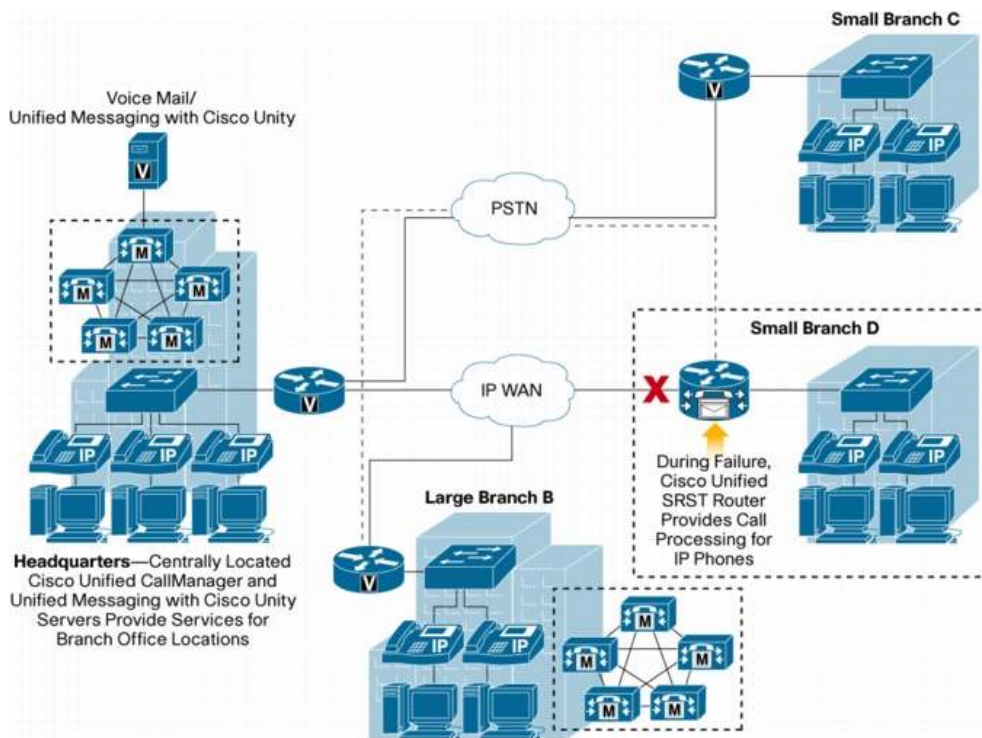
The Cisco Unified Communications system uses Cisco Unified CallManager in combination with Cisco Unified Survivable Remote Site Telephony which is embedded within Cisco IOS® Software, to help provide high-availability IP telephony to branch offices. When access to Cisco Unified CallManager from the branch office is impeded, for example, as a result of a WAN link failure, Cisco Unified Survivable Remote Site Telephony provides telephony backup services to help ensure that the branch office has continuous telephony service over the Cisco network infrastructure deployed in the branch. The enhanced reliability provided by Cisco Unified Survivable Remote Site Telephony makes the Cisco Unified Communications system a cost-effective solution to help ensure telephony operation to all users in an organization, whether they are located in the headquarters or in a branch office.

Furthermore, in certain environments, the security of telephony communication is a critical requirement. The Cisco Unified Communication system supports secure telephony communication between any two phones in the network, whether those phones are in the headquarters facility or at a branch office. Cisco Unified Survivable Remote Site Telephony contributes to this secure telephony communication solution by supporting the same secure telephony protocols in the branch office when the branch has lost communication with the centralized Cisco Unified CallManager.

HOW IT WORKS

Cisco Systems® developed Cisco Unified Survivable Remote Site Telephony technology for all Cisco IOS Software platforms that support voice (refer to Table 3 for a complete list). The Cisco Unified Survivable Remote Site Telephony feature integrates network intelligence into Cisco IOS Software, which acts as the call-processing engine for IP phones located in the branch office during a WAN outage (Figure 1).

Figure 1. Centralized Cisco Unified CallManager Deployment with Remote Site Experiencing WAN Failure; Cisco Router Using Cisco Unified Survivable Remote Site Telephony



Cisco Unified Survivable Remote Site Telephony functions in the branch office router to automatically detect a failure in the network and initiate a process to autoconfigure the router. This provides call-processing backup redundancy for the IP phones in that office and helps to ensure that the telephony capabilities stay operational. Upon restoration of WAN connectivity, the system automatically shifts call processing back to the primary Cisco Unified CallManager cluster. The Cisco Unified Survivable Remote Site Telephony configuration needs to be completed only once during install, simplifying deployment, administration, and maintenance. No IT staff is required at the remote sites to manage the Cisco Unified Survivable Remote Site Telephony feature.

Cisco routers with Cisco Unified Survivable Remote Site Telephony also offer secure voice mode with Cisco Unified Survivable Remote Site Telephony 3.3 and higher. If you deploy secure voice with Cisco Unified CallManager at your main site, secure Cisco Unified Survivable Remote Site Telephony gives you the option to keep calls secure during Cisco Unified Survivable Remote Site Telephony mode with transparent layer security (TLS) and Secure Real-Time Transport Protocol (SRTP) for signaling and media encryption, respectively. When the WAN link or Cisco Unified CallManager is restored, Cisco Unified CallManager resumes secure call-handling capabilities.

Cisco Unified Survivable Remote Site Telephony version 3.4 provides support for Session Initiation Protocol (SIP) for Cisco Unified IP phones, which provides basic telephony functionality when the network SIP proxy or registrar is no longer available. The Cisco Unified Survivable Remote Site Telephony router with SIP enabled provides SIP registrar services during the outage and supports a back-to-back user agent, allowing for supplementary features such as call transfer and forwarding. Cisco Unified IP phones using SIP will register to the Cisco Unified Survivable Remote Site Telephony enabled router when the WAN link is out of service.

Cisco Unified Survivable Remote Site Telephony version 4.0 provides these new features:

- Support for video using Cisco Unified Video Advantage clients
- Support of softphone on Windows PC using Cisco IP Communicator
- Fax pass-through using SCCP for analog telephone adaptors (ATAs)
- Call preservation enhancements between Cisco Unified IP Phone and SRST Gateway using H.323 protocol

Cisco Unified Survivable Remote Site Telephony offers fault monitoring using Simple Network Management Protocol (SNMP) with the SRST-Management Information Base (MIB), which gives you the ability to remotely monitor the Cisco Unified Survivable Remote Site Telephony site using existing SNMP tools or CiscoWorks. The CISCO-SRST-MIB provides the network operations center details about Cisco Unified Survivable Remote Site Telephony activity; including duration of SRST usage, IP phones registered or registration failure, and calls processed during Survivable Remote Site Telephony mode. To receive CISCO-SRST-MIB data to the central site during Survivable Remote Site Telephony mode requires a backup WAN link connection.

Table 2. Cisco Unified Survivable Remote Site Telephony Platform Density and Feature License Part

Platform	Number of Phones Supported***	Part Number	Part Number (Spare)
Cisco 1760-V Modular Access Router and Cisco 2801 Integrated Services Router	Up to 24 phones	FL-SRST-SMALL	FL-SRST-SMALL=
Cisco 2600XM Multiservice and Cisco 2811 Integrated Services Router	Up to 36 phones	FL-SRST-36	FL-SRST-36=
Cisco 2650XM Multiservice Router, and Cisco 2821 Integrated Services Router	Up to 48 phones	FL-SRST-MEDIUM	FL-SRST-MEDIUM=
Cisco 2851 Integrated Services Router	Up to 96 phones	FL-SRST-96	FL-SRST-96=
Cisco 3725 Multiservice Access Router	Up to 144 phones	FL-SRST-144	FL-SRST-144=
Cisco 3825 Integrated Services Router	Up to 336 phones	FL-SRST-336	FL-SRST-336=
Cisco 3745 Multiservice Access Router, Cisco Catalyst® 6500 Series Communication Media Module (CMM)*	Up to 480 phones	FL-SRST-480	FL-SRST-480=
Cisco uBR7200 Series NPE-400 and NPE-G1 Network Processing Engines**	Up to 480 phones	FL-SRST-480	FL-SRST-480=
Cisco 3845 Integrated Services Router	Up to 720 phones	FL-SRST-720	FL-SRST-720=

* The Cisco Catalyst 6500 Series CMM supports Cisco Unified Survivable Remote Site Telephony 3.2 in Cisco IOS Software Release 12.3(8)XY and supports Cisco Unified Survivable Remote Site Telephony 2.1 with Release 12.2(13)ZC.

** The Cisco uBR7200 Series supports Cisco Unified Survivable Remote Site Telephony 2.1 only with Cisco IOS Software Release 12.3 Mainline.

*** The maximum number of SIP phones supported in Survivable Remote Site Telephony mode in this release is different as follows. Cisco 1751, Cisco 1760, Cisco 2610XM, Cisco 2611XM, Cisco 2620XM, Cisco 2621XM and Cisco 2801 - Up to 24 phones; Cisco 2650XM and Cisco 2811 - Up to 36 phones; Cisco 2821 - Up to 48 phones; Cisco 2691 - Up to 72 phones; Cisco 2851 - Up to 96 phones; Cisco 3725 - Up to 144 phones; Cisco 3825 - Up to 168 phones; Cisco 3745 - Up to 192 phones; Cisco 3845 - Up to 480 phones.

CISCO UNIFIED SURVIVABLE REMOTE SITE TELEPHONY PLATFORM INFORMATION

Cisco platforms with Cisco Unified Survivable Remote Site Telephony support from 24 to 720 phones. Details about currently supported platforms and the number of phones per platform is provided in the Cisco Unified Survivable Remote Site Telephony Specifications Sheet for each version, which can be viewed online at:

http://www.cisco.com/en/US/products/sw/voicesw/ps2169/products_documentation_roadmap09186a008018912f.html.

Cisco offers integrated services router bundles with Cisco Unified Survivable Remote Site Telephony at a discount when compared to purchasing bundle components separately. These bundles are listed in Table 3.

Table 3. Cisco Unified Survivable Remote Site Telephony Bundles

Bundle Part Number	Includes
CISCO3845-SRST/K9	Cisco 3845 voice bundle with packet voice digital signal processor (DSP) module (PVDM2-64), Cisco Unified Survivable Remote Site Telephony Feature license for 240 phones, and Cisco IOS SP Services feature set
CISCO3825-SRST/K9	Cisco 3825 voice bundle with packet voice DSP (PVDM2-64), Cisco Unified Survivable Remote Site Telephony feature license for 168 phones, and SP Services
CISCO2851-SRST/K9	Cisco 2851 voice bundle with packet voice DSP (PVDM2-48), Cisco Unified Survivable Remote Site Telephony feature license for 96 phones, and SP Services
CISCO2821-SRST/K9	Cisco 2821 voice bundle with packet voice DSP (PVDM2-32), Cisco Unified Survivable Remote Site Telephony feature license for 48 phones, and SP Services
CISCO2811-SRST/K9	Cisco 2811 voice bundle with packet voice DSP (PVDM2-16), Cisco Unified Survivable Remote Site Telephony feature license for 36 phones, and SP Services
CISCO2801-SRST/K9	Cisco 2801 voice bundle with packet voice DSP (PVDM2-8), Cisco Unified Survivable Remote Site Telephony feature license for 24 users, SP Services
CISCO2651XM-V-SRST	Cisco 2651XM voice bundle with packet voice DSP on AIM-VOICE-30 module, Cisco Unified Survivable Remote Site Telephony feature license for 48 phones, 32-MB flash memory, 96-MB DRAM, and Cisco IOS SP Services feature set
CISCO1760-V-SRST	Cisco 1760 voice bundle with packet voice DSP (PVDM-256K-4), Cisco Unified Survivable Remote Site Telephony feature license for 24 phones, 32-MB flash memory, 64-MB DRAM, and Cisco IOS IP/VOX Plus feature set

CISCO UNIFIED IP PHONE SUPPORT

Cisco Unified Survivable Remote Site Telephony is supported with Cisco Unified CallManager Version 3.01 and greater. Cisco Unified Survivable Remote Site Telephony is not dependent on Cisco Unified CallManager versions but on IP phone loads.

Table 4 lists the Cisco Unified IP phones supported by Cisco Unified Survivable Remote Site Telephony with Skinny Call Control Protocol (SCCP) phone loads.

Table 4. Cisco Unified IP Phone Support Using SCCP

Phone	Cisco Unified Survivable Remote Site Telephony 2.1	Cisco Unified Survivable Remote Site Telephony 3.0	Cisco Unified Survivable Remote Site Telephony 3.1	Cisco Unified Survivable Remote Site Telephony 3.2	Cisco Unified Survivable Remote Site Telephony 3.3	Cisco Unified Survivable Remote Site Telephony 3.4	Cisco Unified Survivable Remote Site Telephony 4.0
Cisco Unified IP Phone 7970G, 7971G-GE	-	-	-	-	X	X	X
Cisco Unified IP Phone 7960G and 7940G	X	X	X	X	X	X	X
Cisco Unified IP Phone 7961G, 7941G, 7961G-GE, 7941G-GE	-	-	-	-	X	X	X
Cisco Unified IP Conference Station 7935	X	X	X	X	X	X	X
Cisco Unified IP Conference Station 7936	-	-	X	X	X	X	X
Cisco Unified IP Phone 7912G	-	X	X	X	X	X	X
Cisco Unified IP Phone 7910G	X	X	X	X	X	X	X
Cisco Unified IP Phone 7905G	-	X	X	X	X	X	X
Cisco Unified IP Phone 7902G	-	X	X	X	X	X	X

Cisco Unified Wireless IP Phone 7920	-	X	X	X	X	X	X
Cisco Unified IP Phone 7914 Expansion Module	X	X	X	X	X	X	X
Cisco VG248 48-Port Analog Phone Gateway	X	X	X	X	X	X	X
Cisco ATA 180 Series Analog Telephone Adaptors	-	-	-	-	-	-	X
Cisco IP Communicator	-	-	-	-	-	-	X
Cisco Unified Video Advantage	-	-	-	-	-	-	X

Table 5 lists the Cisco Unified IP phones supported by Cisco Unified Survivable Remote Site Telephony with Session Initiated Protocol (SIP) phone loads.

Table 5. Cisco Unified IP Phone Support Using SIP

Phone	Cisco Unified Survivable Remote Site Telephony 3.4	Cisco Unified Survivable Remote Site Telephony 4.0
Cisco Unified IP Phone 7970G and 7971G-GE	X	X
Cisco Unified IP Phone 7960G and 7940G	X	X
Cisco Unified IP Phone 7961G, 7941G, 7961G-GE, 7941G-GE	X	X
Cisco Unified IP Conference Station 7935	-	-
Cisco Unified IP Conference Station 7936	-	-
Cisco Unified IP Phone 7912G	X	X
Cisco Unified IP Phone 7911G	X	X
Cisco Unified IP Phone 7910G	-	-
Cisco Unified IP Phone 7905G	X	X
Cisco Unified IP Phone 7902G	-	-
Cisco Unified Wireless IP Phone 7920	-	-
Cisco Unified IP Phone 7914 Expansion Module	-	-
Cisco ATA 180 Series Analog Telephone Adaptors	-	-

CISCO IOS SOFTWARE IMAGE SUPPORT

Table 6 summarizes the correlation between Cisco Unified Survivable Remote Site Telephony version and Cisco IOS Software.

Secure Cisco Unified Survivable Remote Site Telephony is available with Cisco Unified Survivable Remote Site Telephony 3.3 and higher for Cisco Unified IP phones using SCCP and also requires Cisco Unified CallManager 4.1(2) and higher.

Cisco Unified Survivable Remote Site Telephony for SIP phones is supported with Cisco Unified Survivable Remote Site Telephony 3.4 and only with Cisco IP phones.

For the latest Cisco IOS Software release and features, consult the Feature Navigator at: <http://www.cisco.com/go/fn>.

Table 6. Cisco IOS Software Release(s)

Cisco Unified Survivable Remote Site Telephony Version	Cisco IOS Software Release(s)
Cisco Unified Survivable Remote Site Telephony 2.0	12.2(13)T
Cisco Unified Survivable Remote Site Telephony 2.1	12.2(15)T and 12.3 Mainline
Cisco Unified Survivable Remote Site Telephony 3.0	12.3(4)T
Cisco Unified Survivable Remote Site Telephony 3.1	12.3(8)T
Cisco Unified Survivable Remote Site Telephony 3.2	12.3(11)T
Cisco Unified Survivable Remote Site Telephony 3.3 Plus Secure SRST	12.3(14)T or 12.4 Mainline
Cisco Unified Survivable Remote Site Telephony 3.4	12.4(4)T
Cisco Unified Survivable Remote Site Telephony 4.0	12.4T Fourth Release

SUPPORTED FEATURES

Cisco Unified Survivable Remote Site Telephony provides robust support for many IP phone features through the duration of the WAN failure, a feature that is not available from other traditional telephony solutions. Features supported during the failure are listed in Table 7.

Table 7. Cisco Unified Survivable Remote Site Telephony Features

Cisco Unified Survivable Remote Site Version	Feature Set
Cisco Unified Survivable Remote Site Telephony 2.0	<ul style="list-style-type: none"> • Support for IP and analog phones • Re-homing of IP phones upon failure to branch router for call processing • Maintenance of local extension-to-extension calls upon failure* • Maintenance of extension-to-public switched telephone network (PSTN) calls upon failure • Up to six lines per phone • Call hold and pick up • Speed and last-number redial • Up to 24 line appearances per system • Primary line support • Maintenance of existing calls upon recovery • Analog foreign exchange office (FXO) and foreign exchange station (FXS) • Calling-party name • Caller ID and asynchronous network interface (ANI) support • WAN link support: Frame Relay, ATM, Multilink Point-to-Point Protocol (MLPPP), serial, ATM Adaption, Layer 2 (AAL2), and DSL • Class of restriction • Music on hold (MOH), tone on hold, and music and tone on transfer (MOH for endpoints PSTN only) • Distinctive ringing • Direct inward dialing (DID) and direct outward dialing (DOD) • PSTN T1 and E1 channel associated signaling (CAS) trunks support • ISDN Basic Rate Interface (BRI) and Primary Rate Interface (PRI) support • Call-detail recording and RADIUS server • Interworking with Cisco Gatekeeper • Transfer to voice mail pilot number using PSTN • Alias lists for unregistered phones • Translation rules support • Tool Command Language (TCL)-based simple automated attendant and interactive voice response (IVR) on local gateways • Transfer across H.323 network of Cisco endpoints
Cisco Unified Survivable Remote Site Telephony 2.1	<ul style="list-style-type: none"> • Cisco Unified CallManager phone language support • Global call-forwarding enhancement • In-band dual tone multi-frequency (DTMF) voice mail integration • Enhanced dial-plan pattern

Cisco Unified Survivable Remote Site Telephony 3.0	<ul style="list-style-type: none"> • E1-R2 signaling support • Secondary dial tone • Dual line appearance per button • Three-party G711 temporary conferencing • Call transfer with consult • MOH multicast from flash .au file in Cisco Unified CallManager mode • Support for Cisco Unified IP Phone 7905 • European date formats • Enhanced dialplan-pattern command • Increased directory-number maximums • Additional language options for IP phone • Configurable system message • Improved debugs for phones • Symmetric SIP gateway-to-gateway DTMF relay • Ringing timeout for phones • Cisco SIP phone support of basic calls only
Cisco Unified Survivable Remote Site Telephony 3.1	<ul style="list-style-type: none"> • Support for Cisco Unified IP Wireless Phone 7920 • Support for Cisco Unified IP Conference Station 7935 or Cisco Unified IP Conference Station 7396
Cisco Unified Survivable Remote Site Telephony 3.2	<ul style="list-style-type: none"> • Enhancement to the alias command • Enhancement to the cor command • Enhancement to the pickup command • Enhancement to the user-locale command • Increased number of phones supported on the Cisco 3745 Multiservice Access Router • MOH Multicast from live-feed in Cisco Unified CallManager mode • No timeout for call preservation* • RFC 2833 DTMF relay support • Translation profile support
Cisco Unified Survivable Remote Site Telephony 3.3	<ul style="list-style-type: none"> • Support for Cisco Unified IP Phone 7970G, 7971G-GE, 7961G, 7941G, 7961G-GE, 7941G-GE, and 7911G • Enhancement to the show ephone command (new Cisco Unified IP Phone model keywords)
Secure Cisco Unified Survivable Remote Site Telephony 3.3 with Cisco Unified CallManager 4.1(2)	<ul style="list-style-type: none"> • Basic call • Call transfer (consult and blind) • Call forward (busy, no answer, and all) • Shared line (IP phones) • Hold and resume • Hold and pickup • Only secure calls between IP phones and/or Cisco Unified Survivable Remote Site Telephony router
Cisco Unified Survivable Remote Site Telephony 3.4	<ul style="list-style-type: none"> • Fault monitoring with SNMP CISCO-SRST-MIB • Cisco Unified Survivable Remote Site Telephony state and duration • Phone registration and failure • Threshold un-registration • Total calls handled during Cisco Unified Survivable Remote Site Telephony mode • Cisco Unified Survivable Remote Site Telephony support for Cisco Unified IP phones using SIP loads • Cisco Unified Survivable Remote Site Telephony support for Cisco Unified IP Phones using SIP loads • SIP Proxy/registrar services during Cisco Unified Survivable Remote Site Telephony mode plus back-to-back user agent for support of supplementary features • SIP features; call forward, call hold, call transfer (blind/consult), distinctive ringing, time-based call blocking, plus SIP phone load features
Cisco Unified Survivable Remote Site Telephony 4.0	<ul style="list-style-type: none"> • Support of video calls with Cisco Unified Video Advantage Client • Support of Cisco IP Communicator • Fax pass-through using SCCP with Cisco ATA 180 Series Analog Telephone Adaptors • Call preservation enhancements between IP phones and H.323 controlled voice gateways

* Prior to Cisco Unified Survivable Remote Site Telephony 3.2, active calls to the PSTN from Cisco Unified Survivable Remote Site Telephony IP phones are maintained for most calls and dropped after approximately three minutes. Active calls between users on the same LAN are not affected by WAN failure, and security is maintained for the duration of the call.

Cisco Unified Survivable Remote Site Telephony 3.2 and later can preserve existing H.323 calls on the branch if an outage occurs, by disabling the H.225 keepalive timer by entering the no h225 timeout keepalive command.



SERVICE AND SUPPORT

Using the Cisco Lifecycle Services approach, Cisco Systems and its partners offer a broad portfolio of end-to-end services. These services are based on proven methodologies for deploying, operating, and optimizing IP Communications solutions. Upfront planning and design services, for example, can help you meet aggressive deployment schedules and minimize network disruption during implementation. Operate services reduce the risk of communications downtime with expert technical support. Optimize services enhance solution performance for operational excellence. Cisco and its partners offer a system-level service and support approach that can help you create and maintain a resilient, converged network that meets your business needs.

SUMMARY

Cisco Unified Survivable Remote Site Telephony, in combination with Cisco Unified CallManager, offers enterprises a simple, cost-effective solution for customers who want the benefits of a centralized call-processing architecture with redundancy at the remote office.

For more information about the Cisco Unified Communications system, visit the following:

- Cisco Unified Survivable Remote Site Telephony product and technical information: <http://www.cisco.com/go/srst>
- Cisco Unified Communications products, including Cisco Unified CallManager: <http://www.cisco.com/go/voice>



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