# Cisco on Cisco Best Practices
# Survivable Remote Site Telephony Design Guide

# Table of Contents



 

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# 1 Introduction

This document specifies Cisco IT infrastructure standards for Survivable Remote Site Telephony (SRST) to support remote-office IP telephony business continuity for Cisco remote offices. It prescribes the use of SRST technology for consistent architecture, design, and deployment in situations that share similar criteria.

Before you read this document, Cisco recommends that you read the Cisco IT Remote Office Design Guide(http://www.cisco.com/web/about/ciscoitatwork/network_systems/branch_office_network_design.html),which describes Cisco branch office architecture and design.

This design guide describes Cisco IT SRST deployments for single- or dual-router remote offices. The design includes standards for all deployment models as outlined in the centralized call-processing architecture. This design guide also summarizes the base standard configuration for the platform and outlines different solutions depending on the number of devices to be supported in SRST mode. This design guide does not address high-level architecture and design of IP telephony topology, nor does it provide specific IP telephony configuration examples.

# 2 Glossary

The following terms and definitions are used in this document:

**AA** Auto Attendant

**AAR** Automated Alternate Routing

**CCM** Cisco Unified Communications Manager (formerly Cisco CallManager)

**CMM** Communications Media Module. Gateway platform

**COR** Class of Restriction

**DID** Direct Inward Dialing (used with PRI services)

**DSP** Digital Signal Processor

**EMAN** Enterprise Management

**FSO** Field Sales Office

**FXO** Foreign Exchange Office

**FXS** Foreign Exchange Station

**GK** H.323 Gatekeeper with via-zone capabilities

**Greenfield** Completely new site deployment without the requirement to integrate existing systems

**H.323** ITU-T Recommendation: packet-based multimedia communications systems

**ICT** Inter-Cluster Trunk

**IPC** IP Communicator

**IPT** IP telephony environment

**MGCP** Media Gateway Control Protocol (Cisco Unified Communications Manager controlled gateway protocol)

**MoH** Music on Hold

**SCCP** Skinny Client Control Protocol (used by devices to communicate with Cisco Unified Communications Manager)

**SRST** Survivable Remote Site Telephony. Cisco IOS Software feature that allows call-processing functionality during loss of connection to central Cisco Unified Communications Manager cluster.

**TCO** Total Cost of Ownership

**VAD** Voice Activity Detection

**VSO** Very Small Office

# 3 Cisco Remote Office Deployments

Cisco IT supports more than 300 remote sites in approximately 100 countries. Some of these sites are campus sites, but most are remote branch offices. Remote branch offices connect to the campus sites through the Cisco WAN to hub sites. The Cisco campus and hub sites have dedicated Cisco Unified Communications Manager (formerly Cisco CallManager) clusters. Cisco policy provides that remote users receive the same level of services as employees at campus sites. This approach includes unified communications applications such as voice mail, interactive voice response (IVR), and video services. Only conferencing resources must be locally provisioned at the remote site to prevent multiple streams from unnecessarily traversing the WAN.

### Hardware Requirements

The following addresses the use of the network modules slots and Voice/WAN Interface Card (VWIC) slots at Cisco IT. The available voice slots can be provisioned for primary rate interface (PRI), basic rate interface (BRI), FXO, or FXS connections. Voice lines must be scaled to provide sufficient public switched telephone network (PSTN) connectivity for remote offices. Because many variables determine how to size the PSTN connectivity for a site (for example, business function and cost), this issue is addressed on a site-by-site basis. The standard modules necessary for provisioning PSTN connectivity based on site SRST capacity are described later in this document. When defining which deployment model best suits an office to ensure that endpoint quantities fall within these SRST limits, the design addresses sites with no more than 80 percent of the maximum phones supported in SRST across both routers on the Cisco ISR 3845 platform. SRST is configured on one or both routers depending on the number of endpoints to be supported, and with PSTN connectivity divided across both routers where possible. Based on IT remote office standards, this design defines three configurations that can be applied to remote sites with the following criteria:

- **Dual WAN router remote site – Single SRST (configured on a single router)**
  Survivable endpoint total is below 576

- **Dual WAN router remote site – Dual SRST (configured on both routers)** Survivable endpoint total exceeds 576
  Survivable endpoint total is below 1152

- **Single WAN router remote site – Single SRST**
  Survivable endpoint total is below 576

Note: *Figures are based on Cisco IOS Software Release 12.4.3 SRST 3.2 with Cisco ISR 3845 routers as WAN gateways. Consult www.cisco.com for the most recent updates based on latest versions.

In dual router offices, configuring SRST across both routers effectively increases the SRST capacity to 1152 when configured on the Cisco IT standard Cisco ISR 3845 router. This configuration requires splitting a site's devices into two logical groups within Cisco Unified Communications Manager administration with separate SRST references to help load balance the failover of endpoints between the two routers. Using dual Cisco 3845 routers for a large office solution imposes strict limitations, namely PRI capacity and WAN connectivity, which should be investigated prior to deployment. If critical business requirements at a site warrant a dedicated cluster, Cisco IT analyzes the individual requests before approving this design.

**SRST Components**

SRST is a Cisco IOS Software capability that acts as a backup call-processing agent for Cisco Unified Communications Manager and provides a subset of voice capabilities if the connection to the central Cisco Unified Communications Manager cluster fails. SRST is deployed as standard to all remote offices using the centralized call-processing model to help ensure greater resilience and reliability for the supported clients. IOS and hardware dependent, SRST can currently support a maximum of 1440 endpoints, although Cisco IT recommends reducing this amount by 80 percent to account for an anticipated CPU increase introduced by future IOS features. Different deployment models are tailored to suit other site capacities below this figure. Cisco IT's SRST standards for a Cisco remote office recommend that a remote site, at a minimum, must support the following capabilities in SRST:

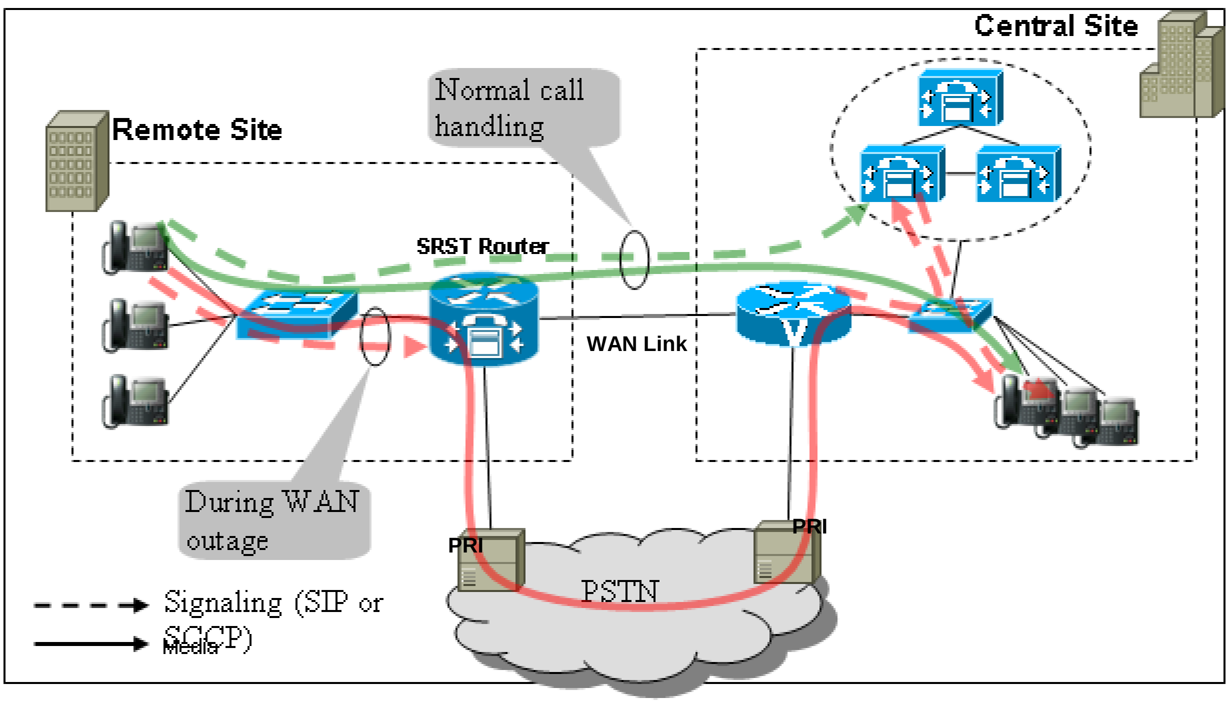
- Provide inbound and outbound PSTN connectivity to all phones at a remote site
- Reroute undefined DID extensions to reception/switchboard
- Provide acceptable PTT calling party CLI on outbound calls
- Non-integrated voice mail over PSTN for messages button
- Call forward busy/no answer to:
  First choice - voice mail (where possible)
  Second choice - reception/switchboard
- Date and time
- Eight-digit translation to PSTN format for internal support services
- MoH locally streamed from SRST router
- Inter-digit and busy timeout values

## 3.1.1 SRST

Under normal operations, when the WAN is available, the centralized Cisco Unified Communications Manager does all of the call routing. At that point, only the POTS dial peers for the PSTN interface (PRI/BRI/FXO) are used for external call routing, and then only for incoming calls from the PSTN, and if present, dial peers for FXS fax stations. E-phone dial peers (dial peers in the SRST routers for phones) are not active, because the phones are not registered with the SRST router. During this time, IP phones at the central and remote site behave in the same way, registering with one of the Cisco Unified Communications Managers at the central site, as determined by the Cisco Unified Communications Manager group configured for each phone (primary, secondary Cisco Unified Communications Managers).

When a WAN outage occurs, Skinny Client Control Protocol (SCCP) phones detect TCP keepalive timeouts (default keepalive intervals 30 seconds) to all configured Cisco Unified Communications Managers, and fail over to SRST by sending a registration for each device to the SRST router. This triggers the SCCP SRST feature (e-phone) to enable POTS dial peers for each directory number registered. It is only at this time that all call routing decisions move to the SRST gateway. Figure 1 shows a basic call from a remote site to the central site before and after a WAN outage. Prior to the outage, all of the call signaling goes through the Cisco Unified Communications Manager (shown in green).

**Figure 1 SRST Call Routing**




During the outage (shown in red), VoIP call signaling terminates at the SRST router, and the call is routed through the PSTN to the central site. The SRST feature becomes active as soon as the phone detects that it has lost its connections to all Cisco Unified Communications Managers, including the ones configured in the redundancy group. At this point the clients will see the default display message for IP phones in fallback mode: “CM Fallback Service Operating.” Calls in progress to local PSTN services are survivable during failover. IP phones periodically attempt to re-establish a connection with the primary Cisco Unified Communications Manager based on response acknowledgments to keepalives at 30-second intervals. When a connection is re-established with the primary Cisco Unified

Communications Manager, Cisco IP phones automatically cancel their registration with the SRST router.

### 3.1.2 Cisco Unified Communications Manager

Cisco Unified Communications Manager is the main call-processing agent for remote site telephony while WAN connectivity between the main, central cluster, and remote site is available. Cisco INS IT opted for the centralized call-processing model to reduce server count and associated total cost of ownership. It provides centralized call-processing functions for remote site devices as well as applications and services such as video, security, and XML applications. This document focuses only on the Cisco Unified Communications Manager configuration that is integral to deploying this SRST standard.

### 3.1.3 Communication Endpoints

All communication endpoints deployed at sites using SRST should have SRST support so that the endpoints can re-register to the SRST routers in the event of a WAN outage. If a certain endpoint does not have SRST support, this limits its application to noncritical uses that can tolerate outages or service unavailability. Cisco IT has standardized on desktop phones and conference room phones with full SRST support.

### 3.1.4 Gateways

Remote office gateway components comprise four main categories:

- Analog station gateways (FXS)
- Digital trunks to PSTN (PRI)
- Digital trunks to PSTN (BRI)
- Analog trunk to PSTN (FXO)

#### 3.1.4.1 Analog Station Gateways (FXS)

The Foreign Exchange Station (FXS) gateway is the interface on a VoIP device for connecting directly to analog devices, such as faxes and modems, and supplies ring, voltage, and dial tone.

These interfaces terminate to FXS modules in the SRST router itself or on a VG224 if the router does not provide sufficient capacity. Cisco IT chooses the module type based on the quantity of analog ports required to support the analog devices at a remote site and the type of site being provisioned.

**Recommended for dual router FSO**; FXS/FXO modules include:

EVM-HD-8FXS/DID
EM-HDA-8FXS
EM-HDA-6FXO
EM-HDA-3FXS/4FXO

These modules can provide up to 24 FXS termination points for analog devices based on using one slot (slot 4) on a single SRST router. This capacity can be doubled using slot 4 in both routers.

Recommended additional FXS device:

VG-224 (24 FXS) – External to WAN router
In sites where the numbers of required FXS devices are greater, and/or recommended cable lengths are exceeded between the WAN routers and FXS device, VG-224 devices can be deployed. This IOS device can support up to 24 FXS devices.

**Recommended for single router VSO**; FXS/FXO modules include:
VIC-4FXS/DID in VWIC slot 4

Note: Max. cable lengths
(100BaseT IEEE 802.3u)
100BaseT4 Category3 or 4 = 100 meters (4 pair)
100BaseTX Category5 = 100 meters (2 pair)
Category 3, 4, or 5 (4 pair) = 100 meters

### 3.1.4.2 Digital Trunks to PSTN (PRI/BRI)

Digital trunks used to connect to PSTN are favored over analog-based devices and fall into two categories: primary rate interface (PRI) and basic rate interface (BRI). PRI is a type of ISDN service designed for larger remote offices. A PRI consists of 30 B-channels (23 in the United States) and one D-Channel, and is transmitted through an E1 (EMEA, APAC, India and Japan) or T1 (United States). For smaller remote sites, a single or multiple BRI is used, which only contains two B-channels and one D-channel. Both gateway types provide features such as DTMF support for dial tone and fax/modem support, and use protocols such as MGCP, H.323, and SCCP for local call processing. In most cases, both the BRI and PRI interfaces have associated Direct Inward Dial (DID) capabilities, allowing internal office extension ranges to be assigned to one or more trunks and therefore creating the ability to dial individual extensions from the PSTN. Where individual DID’s are not available, a single main office DID is required and associated to the trunks that terminate to reception, switchboard, or AA, and then internal NON-DID extensions are accessed through one of these services.

**Field Sales Office dual router recommendations for PRI digital trunk termination.** Cisco IT includes a VWIC-(1 or 2)-MFT-E1 / T1, where T1 is the U.S. standard and E1 covers the rest of the world. These are placed in the main internal motherboard’s VWIC slot 1 or 2, allowing for up to four PRI connections per router. VWICs are provisioned across both routers, regardless of whether PRIs are spread between both devices to provide a redundant module if one fails. For small offices, the BRI module VIC2-2BRI-NT/TE is considered a supported exception. It is rarely required because sites that meet minimum requirements for this design are large enough to warrant either a T1 or E1. If used, they are placed in the main board’s slot 1 or 2 of SRST gateway 2, allowing for up to four BRI connections per router (with additional connections being distributed to gateway 1).

**Very Small Office single router recommendations for PRI/BRI digital trunk termination.** Cisco IT includes VWIC-(1 or 2)-MFT-E1/T1 with 2 x PVDM2-64’s on the motherboard. If needed, the VWIC slots are provisioned with VIC2-BRI-NT/TE and VIC2-4FXO.

#### 3.1.4.3 Analog Trunk to PSTN (FXO)

A Foreign Exchange Office (FXO) gateway is used for trunk or tie-line connections to PSTN, also providing ring, voltage, and dial tone. As a best practice, Cisco IT uses this type of gateway in a remote office environment where BRI or PRI connections are not available. On a single router, the hardware listed above (Analog Station Gateways 4.1.4.1) at this time supports up to 10 FXO connections, although this would only leave eight FXS terminations available. Where required, the EM-HDA-3FXS/4FXO is used for voice backup on gateway 1 and extra FXS capacity, although an additional EVM-HD-8FXS is necessary to house the submodule. Alternatively, a VIC2-4FXO on the main board (slot 1 or 2) of SRST gateway 1 can also accommodate this requirement if additional FXS capacity/redundancy is not required.

In the United States, remote sites are configured with backup voice lines to provide a level of redundancy if the primary voice link fails. The number of redundant lines required depends on the total amount of traffic or users at a location and the physical layout of the site (multiple buildings, floors, contiguous space, etc.). In most cases, four FXO lines are provisioned in sites that share PSTN termination facilities with other companies within the same building. These situations tend not to provide reliable and adequate power redundancy. FXO lines are not dependent on local power and, as such, remain available during a power outage.

### 3.1.5 Media Resources

The following media resources provide supplementary services to remote sites. Some of these resources are delivered at the remote site via local hardware, while others are provisioned centrally at the hub site.

- Conferencing
- MoH
- Transcoding

Local DSP resources are needed in remote sites to provide conferencing capabilities to the IP phones in that location. IP phones can then initiate ad-hoc conference calls and support a limited number of participants (up to six as defined in the service parameters of the Cisco Unified Communications Manager cluster). Hardware conference bridge resources are required to merge and deliver the different voice streams. Using centralized resources for this task would result in voice streams crossing the WAN unnecessarily and therefore consuming a remote site's limited bandwidth. On the Cisco ISR 3845 voice gateway, DSPs for standard voice termination for the voice trunk groups are shared with conferencing resources.

**Single Router Very Small Office – Single SRST**

2 x PVDM2-64's

**Dual Router Field Sales Office – Single SRST (SRST configured on a single router)**
2 x PVDM2-64's provisioned on both routers
**Dual Router Field Sales Office – Dual SRST (SRST configured on both routers)**
4 x PVDM2-64's provisioned on both routers

The dual SRST 3845 must contain four PVDM2-64's installed on the main board per SRST enabled router to provide both conferencing resources and DSP banks to accommodate normal voice termination. DSP provisioning is an extremely complex process, and opting for the highest capacity chip set for the dual SRST model greatly simplifies equipment choice. Also, by over-provisioning in this manner, Cisco IT allows for future office expansion and provides adequate redundancy in case a DSP chip-set fails.

**Figure 2 Cisco ISR 3845 Main Board with 4 x PVDM2 Slots**


In both cases, a Media Resource Group (MRG) is configured per site and is assigned as the first option to the devices on that site. This ensures that local resources are used first, and central resources are used only if local resources are not available,

Note that Cisco IT standardizes on provisioning hardware conference resources only; no software conference bridges are created on Cisco Unified Communications Manager servers.

Because most inter-site voice traffic uses G.729 to enable a more economic use of bandwidth, transcoding resources are provided at the core sites only. Transcoding G729 to G711 is typically required only for centralized application servers (that is, IVRs, Cisco Unity voice mail). Allocating the core sites' DSPs on the WS-X6608-E1/T1 blade provides these resources.

The Music on Hold (MoH) feature provides a music stream to callers who have been put on hold while they are waiting to be transferred to another phone or until the called user is available. The current Cisco call-processing topology provides redundant centralized MoH servers to stream the music to the endpoints using G.711 to provide best audio quality. Currently, the standard best practice is to configure MoH for all remote sites regardless of bandwidth limitations based on the premise that Automated Attendant Routing (AAR) is configured on the cluster. If AAR is not available for a remote site because of a lack of DID functionality or very limited bandwidth, disabling MoH is optional. This is handled on a case-by-case basis.

## 3.2 SRST Deployment Models

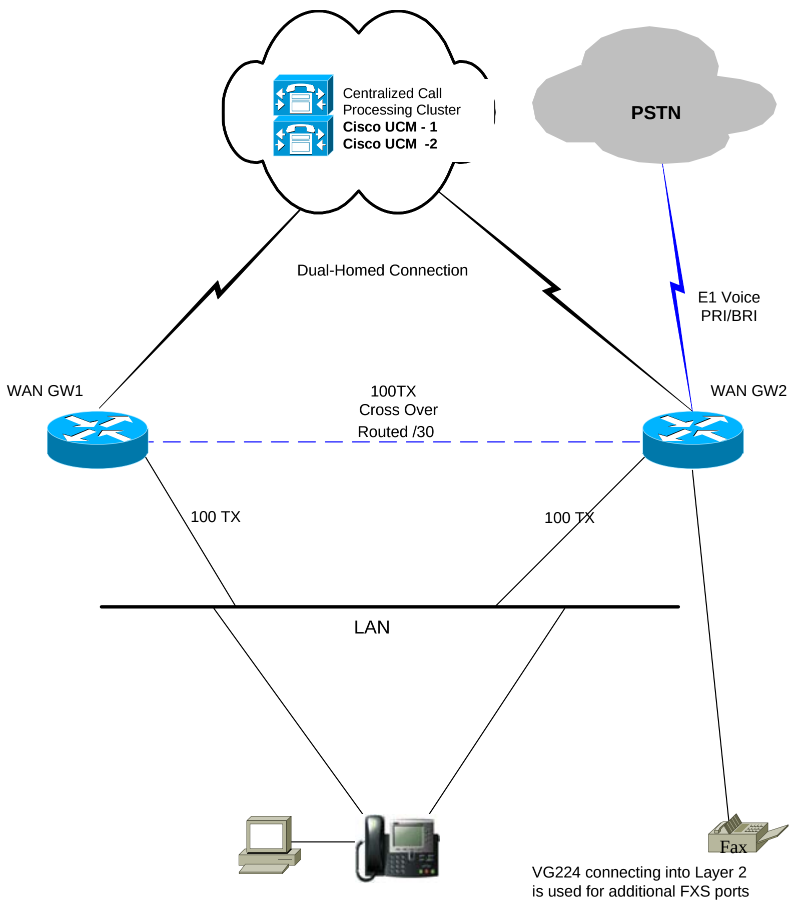
The following network diagrams give examples of the different standard topology types in each global theater. Because the transport infrastructure implementations vary within these theaters, the standard remote site voice models also vary slightly. For detailed deployment model topologies and qualifying criteria, refer to the Cisco IT Remote Office Design Standards at
http://www.cisco.com/web/about/ciscoitatwork/network_systems/branch_office_network_design.html

## 3.2.1 Dual-Homed/Dual Router Site Topology

This network type is based on dual, clear channel leased lines that are sized appropriately to accommodate the remote site. In normal operations, traffic is sent across the primary link and the backup is active only if the primary link fails. PSTN redundancy can be provisioned in numerous ways, depending on the environment where the circuits terminate. Spare VWIC voice modules must be installed on the secondary gateway (gw1) to provide redundancy if the hardware to which the PSTN circuits terminate on gateway 2 fails. In this scenario, the circuit can easily be repatched to gateway 1. Repatching in this way, although not desirable, is the method that has the fewest impacts, is the most cost effective, and requires minimal skill level. The only other option is PSTN over provisioning, which is cost prohibitive because of the very low risk of hardware failure. Allocating audio and video bandwidth in line with the QoS design recommendations, blocking SCCP traffic on the backup link if it is smaller than the primary link to prevent oversubscription if the primary link fails are other best practices followed for this design.

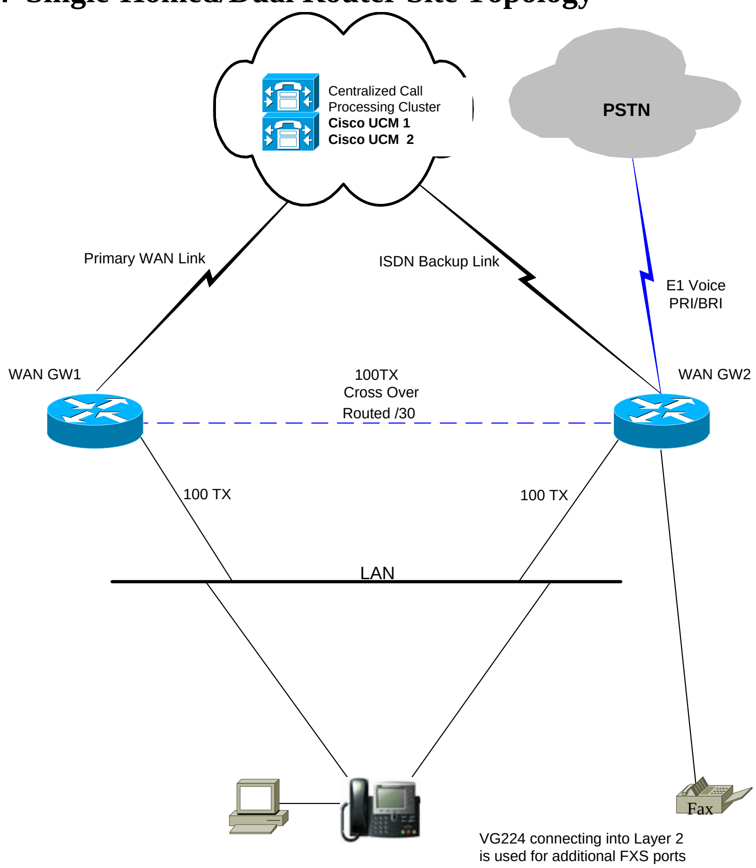
**Figure 3 Dual-Homed/Dual Router Site Topology**



## 3.2.2 Single-Homed/Dual Router Site Topology

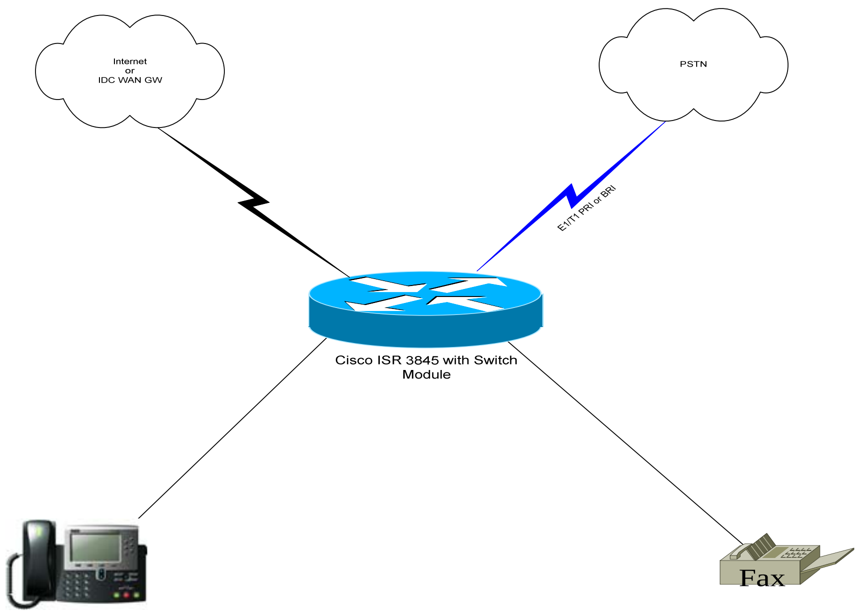
This topology is used for locations where ISDN backup is still predominantly used for a backup link and the backup is far smaller than that of the primary. Some of the best practices followed include -- Allocating audio and video bandwidth in line with the QoS design standards, blocking voice and video traffic on the backup link based on the size of the largest single link and installing spare VWIC voice modules on the secondary gateway 1.

**Figure 4 Single-Homed/Dual Router Site Topology**



### 3.2.3 Single-Homed/Single Router Topology (Global)




## 3.3 SRST Component Configuration

### 3.3.1 SRST (IOS) Configuration

The following commands outline the standard SRST features, function, and usage as applied inside Cisco IT. The commands are based on the following IOS and SRST version:

Cisco IOS Software Release 12.4.3

SRST version 3.2

| IOS cmd | example | Comment/Usage |
|---|---|---|
| From global configuration mode, enter the following commands: | | |
| **Translation-rule 1** **Rule 1** < Internal Support number) if required> <fully qualified PSTN equivalent number> | US SRST COnfiguration– rule 1 12345678 94081234567 | Maintains Internal support speed-dial on IP phone by replacing 8-digit Internal Technical Support speed-dial with PSTN-qualified equivalent in SRST conditions. |

| | | |
|---|---|---|
| **Translation-rule 2**<br>**Rule 1** 8XYZ <PSTN prefix of the remote site><br>(Where XYZ represents the remote sites 3 internal prefix) | Rule 2 840812345678 | Strips 8 plus 3 digit site code from calling number on calls to PSTN and replaces with DID prefix to present recognizable digits to preserve calling party CLI. Modify rule to suit PTT requirement. |
| **Call-manager-fallback** | | Enters CallManager fallback mode |
| **From the call-manager-fallback configuration mode. enter the following commands:** | | |
| **ip source-address** <interface Address> **port 2000 strict-match** | | Usually configured to point to the loopback address |
| **max-ephones** <maximum number of phones to be supported on SRST mode> | Max-ephones 576 | Sets maximum number of endpoint's to be supported in SRST mode. 3845–576 |
| **max-dn** <maximum number of lines, to be double the amount of phones> | Max-dn 960 | Sets maximum number of directory numbers to be supported in SRST mode. 3845–960 |
| **huntstop** | | Sets the huntstop attribute for the dial peers associated with the Cisco IP phone dial peers created during CallManager fallback.<br>Prevents dial-peer hunting. |
| **Translate called** <1> | Translate called 1 | Applies translation rule to Internal Technical support speed-dial (see above) |
| **translate calling** <2> | Translate called 2 | Applies translation rule to calling party number (see above) |
| **voicemail** <Remote voicemail number including outside access code> | Voicemail 4081234567# | Adds PSTN qualified voice-mail number to be assigned to the messages button in SRST mode.<br>(To speed up the dialing of the number add a # at the end of the string.) |
| **moh music-on-hold.au** | | Provides Music on Hold in SRST mode. (File must be stored in Flash) |
| **time-format** <24> | **time-format** 24 | Provides standard 24-hour time display on endpoint in SRST mode. |
| **Date-format** dd-mm-yy | **date-format** dd-mm-yy | Provides standard date format display on endpoint in SRST mode. |
| **Call-forward busy** 8XYZ<voicemail/receptionist's extension> | call-forward busy 81234567 | Sets call fwd all to reception number NOTE: When forwarding to central voicemail system the Redirecting Number IE Delivery--Inbound must be checked on the centralized gateways for access to Cisco Unity via the PSTN during fallback. |
| **call-forward noan** 8XYZ<voicemail/receptionist's extension> timeout 12 | call-forward noan 81234567 | Sets call fwd noan to reception number. NOTE: When forwarding to central voicemail system the Redirecting Number IE Delivery--Inbound must be checked on the centralized gateways for access to the Unity via the PSTN during fallback. |

| | | |
|---|---|---|
| **Transfer-pattern** <outside access code>T | transfer-pattern 0T | Enables PSTN access in SRST<br><br>*transfer-pattern*—String of digits for permitted call transfers. Wildcards are permitted. |
| **Transfer-pattern** <4 digit extension range> | transfer-pattern 10.. | Enables 4-digit dialing in SRST<br><br>*transfer-pattern*—String of digits for permitted call transfers. Wildcards are permitted. |
| **Timeouts interdigit** <8> | Timeouts interdigit 8 | |
| **Timeouts busy** <12> | Timeouts busy 12 | |
| **alias 1** <range of did numbers in the office> to <reception number> | alias 1 81234… 81234567 | Transfers calls to undefined numbers within office DID range to switchboard. command obsoletes the default-destination command and should be used in preference.<br>NOTE: not used when configuring SRST across dual routers. |
| **Limit-dn 7960 <1>** | Limit-dn 7960 1 | Necessary only in countries to prevent VoIP-only line registering to the SRST router. (for example, a legal requirement in India) |
| **Limit-dn 7970 <1>** | Limit-dn 7960 1 | Necessary only in countries to prevent VoIP-only line registering to the SRST router (for example, a legal requirement India). |

## 3.3.2 IOS Gateways

### 3.3.2.1 H.323 Gateway IOS

- Contains standard configuration for H.323 gateway in IOS

| IOS cmd | example | Comment/Usage |
|---|---|---|
| From global configuration mode, enter the following commands: | | |
| **isdn switch-type** <type of ISDN switch> | isdn switch-type primary-net5 | Sets switch type, check with local PTT. |
| **voice class codec 1**<br>codec preference 1 g711alaw<br>codec preference 2 g729r8<br>codec preference 3 g711ulaw | | Sets codec handling<br>NOTE: The order varies by region. Order of codecs should match that of the region where the site is located. |
| **voice class h323 1**<br>**h225 timeout tcp establish** <2> | | Sets timeout for dial peer failover, values can vary. Default value is 5 |
| **interface Loopback**<0 or 5><br>**h323-gateway voip interface**<br>**h323-gateway voip bind srcaddr** <ip address of loopback 0 or 5> | | Sets loopback address as source interface for H.323 signaling.<br>NOTE: Select 0 for APAC and US. Select 5 for EMEA as MPLS SNMP is monitored on loopback 0. |
| To set the controller E1: | | |
| **Controller E1** <slot/port> | controller E1 1/0 | Sets controller for PRI/ fractional PRI |
| **pri-group timeslots** <timeslots> | pri-group timeslots 1-31 | For a full PRI enter E1 1-31as timeslots. |

| | | |
|---|---|---|
| | | For a fractional PRI define the first and last timeslot in use. |
| **Framing** <CRC4 or Non-CRC4> | Framing CRC4 | Sets Framing for PRI. Check with local PTT for details. |
| To set the controller T1: | | |
| **Controller T1** <slot/port> | controller T1 1/0 | Sets controller for PRI/ fractional PRI |
| **pri-group timeslots** <timeslots> | pri-group timeslots 1-24 | For a full PRI enter T1 1-24 as timeslots. a fractional PRI define the first and last timeslot in use. |
| **framing** <esf> | | Check with carrier. Configure when framing loss seconds are increasing. Ensure the framing format configured on the port matches the framing format of the line. Look for "Framing is {ESF\|SF}" in the **show controller t1** output. |
| linecode <b8zs> | | Check with carrier. Configure when line code violations are increasing. Ensure the line coding configured on the port matches the line coding of the line. Look Line Code is {B8ZS\|AMI} in the **show controller t1** output. |
| **no yellow generation**<br>**no yellow detection** | | Check with carrier. Disables yellow detection and generation in the event of frame loss. |
| To set a BRI interface: | | |
| **interface BRI**<slot/module><br>no ip address | interface BRI<slot/module><br>no ip address | |
| **description** < voice/data type circuit id> | Voice BRI PTT052 | |
| **isdn incoming-voice voice** | | Accept inbound ISDN voice calls from PSTN. |
| **isdn send-alerting** | | Check with carrier. Some countries may require an alerting message to be sent before sending the connect message. |
| **isdn sending-complete** | | Check with carrier. Some countries may require a sending complete IE to be sent in the setup message to indicate the entire number is resolved. |
| To set a PRI interface: | | |
| **interface Serial**<br><module/port:channel><br>no ip address | interface<br>Serial<module/port:15><br>no ip address | |
| **description** <voice/data type circuit id> | | See above. |
| **isdn incoming-voice voice** | | See above. |
| **isdn send-alerting** | | See above. |
| **isdn sending-complete** | | See above. |
| To configure voice port: | | |
| **voice-port** <slot/port:channel> | voice-port 1/0:15 | |
| **compand-type a-law**<br>(Non US BRI only) | | Optional. Cisco IOS will default the PCM companding on BRI voice ports to u-law. If the VIC-2BRI-NT/TE is used in locations outside of North America, a-law PCM coding must be selected. NOTE: For PRI this command is not required. |

| | | |
|---|---|---|
| **cptone** <country> | cptone Russia | Specifies a regional analog voice-interface-related tone, ring, and cadence setting. |
| To configure PSTN access: | | |
| **dial-peer voice** 10 pots | | Sets dial-peer identifier for PSTN access. |
| **description**-voice BRI/PRI <e.164 number/circuit id> | description-voice PRI 31203571xxx/CID WOD41684 | Sets assigned interface type, DID range, and circuit id to help troubleshooting. |
| **preference** 1 | | Sets dial-peer preference for the PSTN interface. Preference for subsequent PSTN interfaces should be increased by an incremental value of 1. |
| **destination-pattern** <PSTN access code>T | destination-pattern 0T | Sets destination pattern for external PSTN access utilizing the remote sites PSTN access code. |
| | | |
| **progress_ind alert enable 8** | | Test with carrier. Necessary if voice call send-alert doesn't provide ringback. |
| **progress_ind progress enable 8** | | Test with carrier. Necessary if voice call send-alert doesn't provide ringback. |
| **progress_ind connect enable 8** | | Test with carrier. Necessary if voice call send-alert doesn't provide ringback. |
| **direct-inward-dial** | | |
| **port** <mod/port/channel> | port 1/0:15 | Associates POTS dial-peer to PSTN interface |
| To configure VoIP dial-peer to connect to primary/secondary Cisco Unified Communications Manager - requires 2 VoIP dial-pe | | |
| **dial-peer voice** 200 voip | | Sets dial-peer label for Cisco Unified Communication Manager access. VoIP peer id numbers should start at 200, with subsequent peers increasing by a value of 1. |
| **preference** 1 | | Preference of VoIP dial-peer for primary Cisco Unified Communication Manager subscriber. Preference for subsequent secondary Cisco Unified Communication Manager subscribers should be increased by an incremental value of 1 if destination patterns match. |
| **destination-pattern** 8XYZ…. | destination-pattern 81234… | Sets destination pattern for inbound access utilizing the remote sites 8+7 digit number range.<br>NOTE: Splitting of ranges between clusters requires the creation of extra VoIP dial peers with more specific matches. |
| **incoming called-number** <.> | | Applied to VoIP dial-peers to stop outbound calls from Cisco Unified Communications Manager to router matching the default 'hidden' dial-peer that enables VAD.<br>VAD should be disabled on both inbound and outbound call legs. |
| **voice-class codec** 1 | | Applies codec handling to inbound VoIP peer, defined above. |
| **voice-class h323** 1 | | Applies timeout settings for dial-peer |

| | | |
|---|---|---|
| | | failover, defined above. |
| **session target ipv4:**<ip address of the primary Cisco Unified Communications Manager in the cm group> | | Defines target destination to Primary Cisco Unified Communications Manager on first VoIP peer and secondary Cisco Unified Communications Manager on the second. |
| **dtmf-relay h245-alphanumeric** | | Helps relay DTMF tones without distortion created by Voice codec's. The "h245-alphanumeric" option simply relays DTMF tones as ASCII characters. |
| **tone ringback alert-no-PI** | | Test with carrier. Generates automatic ringback for the caller when no Progress Indicator (PI) alert has been received over the H.323 network. |
| **progress_ind setup enable 3** | | Test with carrier. Forces the gateway to generate ringback.<br>NOTE: Can be used if ringback is not generated due to ISDN issue where ringback is not generated when calls originate from an international location. Ringback for international calls is usually generated on the terminating device. |
| **ip qos dscp** ef media | | Sets correct QoS value for RTP stream on inbound calls. |
| **ip qos dscp** cs3 signaling | | Sets correct QoS value for call signaling on inbound calls. |
| **fax rate** <12000 \| 14400 \| 2400 \| 4800 \| 7200 \| 9600> <disable \| voice> [bytes bytes] | fax rate voice | Sets maximum fax transmission speed. Voice option specifies highest fax transmission rate allowed by the voice codec rate. |
| **no vad** | | Turns off VAD |
| To configure number expansion on the router: | | |
| **num-exp** <range sent by PTT> <8digit sitecode range> | num-exp 123456... 567891.. | Expands digits received from PSTN to 8-digit DID number. |
| **num-exp** <4/5 digits range of the office> <8XYZ….> | num-exp 1… 83121… | Expands 4/5 digit dialing to 8-digit endpoint in SRST. If 4/5 digits are received from PSTN then this line becomes redundant. |
| Optional dial-peer configuration if SRST is configured across both routers.<br>Configure dial-peers on both routers as required.: | | |
| **dial-peer voice** 202 voip | | See above. |
| **preference** 3 | preference 3 | Preference must be a higher number than normal VoIP peers. |
| **destination-pattern** 8XYZ…. | destination-pattern 812345… | Sets generic remote site pattern to match to undefined endpoint dial-peers on originating router in SRST mode. |
| **incoming called-number .**<br>**voice-class codec 1**<br>**voice-class h323 1** | | See above. |
| **session target ipv4:** <ip address of Loopback 0 interface of other Gw> | | Defines target destination to send calls to undefined endpoint dial-peers on originating router in SRST mode. |

| | | |
|---|---|---|
| **dtmf-relay h245-alphanumeric**<br>**ip qos dscp ef media**<br>**ip qos dscp cs3 signaling**<br>**fax rate voice**<br>**no vad** | | See above. |
| Optional dial-peer configuration if PSTN access is configured across both routers<br>Configure dial-peers on both routers as required: | | |
| **dial-peer voice** 507 voip | | See above. |
| **preference** 4 | preference 4 | Preference must be a higher number than existing POTS peers residing on the router itself unless router contains no PSTN interfaces. For example, SRST configured across both routers, but PSTN access is not. |
| **destination-pattern** <PSTN access code>T | destination-pattern 0T | |
| **incoming called-number .**<br>**voice-class codec 1**<br>**voice-class h323 1** | | See above. |
| **session target ipv4:** <Ip address of Loopback 0 interface of other Gw> | | Defines target destination of the other router to increase PSTN capacity. |
| **dtmf-relay h245-alphanumeric**<br>**ip qos dscp ef media**<br>**ip qos dscp cs3 signaling**<br>**fax rate voice**<br>**no vad** | | See above. |

NOTE: Configure ringback tone on ISDN-H.323 VoIP call:
If no ringback tone is heard, refer to the Cisco.com document: Troubleshooting No Ringback Tone on ISDN-VoIP (H.323) Calls

### 3.3.2.2 FXS Gateway IOS

- Contains standard configuration for FXS gateways on a Cisco IOS router

| IOS cmd | example | Comment/Usage |
|---|---|---|
| From global configuration mode, enter the following commands: | | |
| **Dial-peer voice** pots 100 | | Numeric dial-peer identifier, starting at 100 each subsequent FXS peers should increment by a value of 1. |
| **Description** <device type and location> | Description fax M.5-5.7 | Device type and location helps troubleshooting. |
| **Destination-pattern** <8XYZXXXX> | 81234567 | FXS devices 8-digit DN. Only faxes should be assigned a DID number if possible. In no circumstances should a modem be assigned a DID DN. |
| **Port** <x/x/x> | Port 2/0/0 | FXS interface |

| IOS cmd on VG224 | example | Comment/Usage |
|---|---|---|
| From global configuration mode, enter the following commands: | | |
| **voice class codec 1** | | Sets codec handling |
| codec preference 1 g711alaw | | NOTE: The order varies by region. Order of codecs should match that of the region where the site is located. |

| | | |
|---|---|---|
| codec preference 2 g729r8 | | |
| codec preference 3 g711ulaw | | |
| **voice class h323 1** | | Sets timeout for dial peer failover, values vary. Default value is 5. |
| **h225 timeout tcp establish** <2> | | |
| **interface FastEthernet0/0**<br>**description** | | Connection to switch |
| **ip address** <ip address from voice vlan> | | |
| **h323-gateway voip interface** | | |
| **h323-gateway voip bind srcaddr** <ip address of FastEthernet interface><br>duplex full<br>speed 100 | | |
| **interface FastEthernet0/1**<br>**description** | | |
| **No ip address** | | |
| **ip route 0.0.0.0 0.0.0.0 <hsrp address vo: vlan>** | | To be able to send packets out |
| To configure voice port: | | |
| **voice-port** <2/port> | voice-port 2/0 | |
| **cptone** <country> | cptone Russia | Specifies a regional analog voice-interface-related tone, ring, and cadence setting. |
| To configure VoIP dial-peer to connect to primary/secondary Cisco Unified Communications Manager: | | |
| **dial-peer voice** 450 voip | | Sets dial-peer label for Cisco Unified Communication Manager access. |
| **preference** 1 | | Preference of VoIP dial-peer for primary Cisco Unified Communications Manager subscriber. Preference for subsequent secondary Cisco Unified Communications Manager subscribers should be increased an incremental value of 1 if destination patterns match. |
| **destination-pattern** 8XYZ…. | destination-pattern 81234… | Sets destination pattern for inbound access utilizing the remote sites 8+7 digit number range.<br>NOTE: Splitting of ranges between Clusters requires the creation of extra VoIP dial peers with more specific matches. |
| **incoming called-number** <.> | | Applied to VoIP dial-peers to stop outbound calls from Cisco Unified Communications Manager to router matching the default 'hidden' dial-peer that enables VAD.VAD should be disabled on both inbound and outbound call legs. |

| | | |
|---|---|---|
| **voice-class codec** 1 | | Applies codec handling to inbound VoIP peer, defined above. |
| **voice-class h323** 1 | | Applies timeout settings for dial-peer failover, defined above. |
| **session target ipv4:**<ip address of the primary Cisco Unified Communications Manager in the cm group> | | Defines target destination to primary Cisco Unified Communications Manager first VoIP peer and secondary Cisco Unified Communications Manager on the second. |
| **dtmf-relay h245-alphanumeric** | | Helps relay DTMF tones without distortion created by voice codecs. The "h245-alphanumeric" option simply relays DTMF tones as ASCII characters. |
| **ip qos dscp** ef media | | Sets correct QoS value for RTP stream on inbound calls. |
| **ip qos dscp** cs3 signaling | | Sets correct QoS value for call signaling on inbound calls. |
| **fax rate** <12000 \| 14400 \| 2400 \| 4800 \| 7200 \| 9600> <disable \| voice> [bytes bytes] | fax rate voice | Sets maximum fax transmission speed. Voice option specifies highest fax transmission rate allowed by the voice codec rate. |
| **no vad** | | Turns off VAD |
| dial-peer configuration for SRST (remote sites only): | | |
| **dial-peer voice** 202 voip | | See above. |
| **description** | | |
| **preference** 3 | preference 3 | Preference must be a higher number than normal VoIP peers. |
| **destination-pattern** 8XYZ…. | destination-pattern 83121… | Sets generic remote site pattern to match calls to undefined endpoint dial-peers on originating router in SRST mode. |
| **voice-class codec 1** | | |
| **voice-class h323 1** | | |
| **session target ipv4:** <ip address of Loopback 0/5 interface of SRST Gw> | | Defines target destination to send calls to undefined endpoint dial-peers on originating router in SRST mode. EMEA loopback 5; Rest of world – loopback 0 |
| **dtmf-relay h245-alphanumeric** | | See above. |
| **ip qos dscp ef media** | | |
| **ip qos dscp cs3 signaling** | | |
| **fax rate voice** | | |
| **no vad** | | |
| Optional dial-peer configuration if SRST is configured across both routers (remote sites only): | | |
| **dial-peer voice** 203 voip | | See above. |

| | | |
|---|---|---|
| **description** | | |
| **preference** 4 | preference 4 | Preference must be a higher number than normal VoIP peers. |
| **destination-pattern** 8XYZ…. | destination-pattern 83121… | Sets generic remote site pattern to match calls to undefined endpoint dial-peers on originating router in SRST mode. |
| **voice-class codec 1** | | |
| **voice-class h323 1** | | |
| **session target ipv4:** <ip address of Loopback 0/5 interface of other SRST Gw> | | Defines target destination to send calls to undefined endpoint dial-peers on originating router in SRST mode. (EMEA – loopback 5; Rest of world – loopback 0) |
| **dtmf-relay h245-alphanumeric** | | See above. |
| **ip qos dscp ef media** | | |
| **ip qos dscp cs3 signaling** | | |
| **fax rate voice** | | |
| **no vad** | | |
| To configure PSTN access, first via primary/secondary Cisco Unified Communication Manager then PSTN router: | | |
| **dial-peer voice** 204 voip | | Sets dial-peer identifier for PSTN access. |
| **description** | | |
| **preference** 1 | | Sets dial-peer preference for the PSTN interface. Preference for subsequent PSTN interfaces should be increased by an incremental value of 1. |
| **destination-pattern** <PSTN access code>T | destination-pattern 0T | Sets destination pattern for external PSTN access utilizing the remote sites PSTN access code. |
| **voice-class codec 1** | | |
| **voice-class h323 1** | | |
| **session target ipv4:**<ip address primary CCM> | | Defines target destination to primary Cisco Unified Communications Manager on first VoIP peer and secondary Cisco Unified Communications Manager on the second. PSTN GW as third (EMEA – loopback 5 if WAN router; Rest of world loopback 0) |
| **dtmf-relay h245-alphanumeric** | | See above. |
| **ip qos dscp ef media** | | |
| **ip qos dscp cs3 signaling** | | |
| **fax rate voice** | | |
| **no vad** | | |
| **dial-peer voice** 206 voip | | Sets dial-peer identifier for PSTN access. |
| **description** | | |

| | | |
|---|---|---|
| **preference** 3 | | Sets dial-peer preference for the PSTN interface. Preference for subsequent PSTN interfaces should be increased by an incremental value of 1. |
| **destination-pattern** <PSTN access code>T | destination-pattern 0T | Sets destination pattern for external PSTN access utilizing the remote sites PSTN access code. |
| **voice-class codec 1** | | |
| **voice-class h323 1** | | |
| **session target ipv4:**<ip address of Loopback 0/2 interface of the primary PSTN GW> | | Defines target destination of the other router to increase PSTN capacity. |
| **dtmf-relay h245-alphanumeric** | | See above. |
| **ip qos dscp ef media** | | |
| **ip qos dscp cs3 signaling** | | |
| **fax rate voice** | | |
| **no vad** | | |
| Optional dial-peer configuration if PSTN access is configured across both voip routers: | | |
| **dial-peer voice** 207 **voip** | | See above. |
| **description** | | |
| **preference** 4 | preference 4 | |
| **destination-pattern** <PSTN access code>T | destination-pattern 0T | |
| **voice-class codec 1** | | |
| **voice-class h323 1** | | |
| **session target ipv4:**<Ip address of Loopback 0/2 interface of secondary PSTN Gw> | | Defines target destination of the other PSTN router to increase PSTN capacity. |
| **dtmf-relay h245-alphanumeric** | | See above. |
| **ip qos dscp ef media** | | |
| **ip qos dscp cs3 signaling** | | |
| **fax rate voice** | | |
| **no vad** | | |
| To configure the FXS devices with POTS dial-peers: | | |
| **Dial-peer voice** 100 **pots** | | Numeric dial-peer identifier, starting at 100 each subsequent FXS peers should increment by a value of 1. |
| **Description** <device type and location> | Description fax M.5-5. | Device type and location helps troubleshooting. |
| **Destination-pattern** <8XYZXXXX> | 81234567 | FXS devices 8 digit DN. Only faxes should be assigned a DID number if possible. In no circumstances should a modem be assigned a DID DN. |
| **Port** <2/x> | 2/0 | FXS interface |

| For remote sites (non campus) configuration on SRST/PSTN routers to enable VG224 in SRST and for local incoming calls: | | |
|---|---|---|
| **dial-peer voice** 300 **voip** | | Dial peer for analog device connected to VG224 |
| **description** to vg224 80529824 | | |
| **destination-pattern 80529824** | | |
| **session target ipv4:**<IP address of f0/0 interface of VG224> | session target ipv4:1.1.1.1 | VG224 f0/0 interface IP Address |

#### 3.3.2.2.1 FXO Gateway IOS – US Only

- Contains standard configuration for FXO (analogue trunk to PSTN) gateways in IOS

| IOS cmd | example | Comment/Usage |
|---|---|---|
| From global configuration mode, enter the following commands: | | |
| **voice-port** <slot/port> | Voice-port 0/1/1 | |
| **no snmp trap link-status** | | Disables SNMP link trap generation. |
| **connection plar** <reception number> | connection plar 81234567 | Converts inbound calls to FXO line DIDs (not associated to office's main DID range) into main reception number, therefore matching the site's main DID range. |
| **description** <DID number associated to line> | description DIDs 123-456-1234, 4567 | |

## 3.3.3 Media Resources IOS

### 3.3.3.1 Conferencing IOS

| IOS cmd | example | Comment/Usage |
|---|---|---|
| From global configuration mode, enter the following commands: | | |
| **sccp local** <interface-type interface-number> | sccp local GigabitEthernet4/0.400 | Sets the local interface that SCCP applications use to register to Cisco Unified Communications Manager |
| **sccp ccm** <primary ccm> **identifier 1 version x.y** | sccp ccm 10.1.1.1 identifier 1 version x.y | Adds primary Cisco Unified Communications Manager subscriber to the list of available servers to which the Cisco voice gateway can register. |
| **sccp ccm** <secondary ccm> **identifier 2 version x.y** | sccp ccm 10.1.1.2 identifier 2 version x.y | Adds secondary Cisco Unified Communications Manager subscriber to the list of available servers to which the Cisco voice gateway can register. |
| **sccp** | | Enables SCCP and brings it up administratively. |
| **Voice-card** <slot> | Voice-card 2 | Enters voice card configuration mode on the module where DSP farm services are required. |
| **Dsp services dspfarm** | | Enables DSP farm services for the voice card. |
| **exit** | | |

| | | |
|---|---|---|
| **Dsp profile** <profile identifier> <service type> | Dsp profile 1 conference | Enters DSP farm profile configuration mode to define a profile for DSP farm services.<br><br>NOTE: The *profile-identifier* and service type uniquely identifies a profile. If the service type and *profile-identifier* pair is not unique, you are prompted to choose a different *profile-identifier*. |
| **description** <local conferencing> | Description local conferencing | |
| **codec** <codec type> | codec g711ulaw | Specifies the codecs supported by the DSP farm.<br>Repeat this step for each supported codec. |
| **codec** <codec type> | codec g711alaw | |
| **codec** <codec type> | codec g729ar8 | |
| **codec** <codec type> | codec g729abr8 | |
| **codec** <codec type> | codec g729r8 | |
| **codec** <codec type> | codec g729br8 | |
| **maximum sessions** <value> | maximum sessions 6 | Specifies the maximum number of sessions supported by the DSP farm profile for 3845 platform. |
| **dspfarm confbridge maximum sessions** <7> | dspfarm confbridge maximum sessions 7 | Specifies the maximum number of sessions supported by the DSP farm profile for 3745 platform. |
| **associate application SCCP** | | Associates the SCCP to the DSP farm profile. |
| **No shutdown** | | Enables the profile, allocates DSP farm resources, and associates the application. |
| **exit** | | |
| **sccp ccm group** <1> | | Creates a unique Cisco Unified Communication Manager group and enters SCCP Cisco Unified Communication Manager configuration mode. |
| **Description** <cm group> | Description aaagroup | Cisco Unified Communication Manager group |
| **associate ccm 1 priority 1** | associate ccm 1 priority 1 | Associates primary Cisco Unified Communication Manager server to the Cisco Unified Communication Manager group and establishes its priority within the group. |
| **associate ccm 2 priority 2** | associate ccm 2 priority 2 | Associates secondary Cisco Unified Communication Manager server to the Cisco Unified Communication Manager group and establishes its priority within the group. |
| **associate profile** <profile identifier> **register** <device-name> | associate profile 1 register ABCDEFGHIJK | Associates a DSP farm profile to the CallManager group.<br>NOTE: Device-name must match the device name configured in Cisco Unified Communication Manager or profile will |

| | | |
|---|---|---|
| | | not register to Cisco Unified Communication Manager. |
| | | |

# 3.4 SRST Implementation and Operation

## 3.4.1 SRST Implementation Considerations

When establishing how to implement SRST and determine which cluster a remote site should be registered to, Cisco IT considers several factors:

- Closest network path
- Cultural considerations
- High availability services
- Political restrictions
- Legal restrictions
- Cluster sizing

This document assumes a fault-tolerant, highly available network infrastructure with end-to-end QoS for both voice and video traffic. The ITU G.114 recommendation states that the one-way delay in a voice network should be no more than or equal to 150 milliseconds. Edge cases or exceptions that slightly falling outside these recommendations are addressed on a case-by-case basis by the Cisco IT teams.

Cultural considerations include the normal work hours per week within a particular country. For example, the standard work week and weekend days may differ in some countries. This can cause issues when major outages are required at hub cluster locations (servicing remote sites in other countries) for upgrades, patching, etc. where outages are preferred on weekends to prevent unnecessary client disruption.

High-availability, critical services at a remote site might require a standalone cluster or the migration of those services to a central hub cluster location.

Political or military conditions can lead to inadequate network connections or legal restrictions to prevent communication channels. In these cases, remote registration must not be considered, and alternatives such as separate clusters must be investigated.

Many countries have yet to establish a legal framework to govern VoIP communication; therefore, it is essential that the Cisco legal department confirms a country’s legal status before implementing VoIP. In countries where voice signaling is legal but VoIP is not, it is possible to register remote sites on the strict condition that VoIP is blocked with appropriate calling between sites rerouted over PSTN.

Cisco Unified Communications Manager has hardware limitations governing the total device weight that can be loaded onto the platform; this weight must be considered before registering remote sites. It is prudent to keep a running total of the device/dialplan weight on a cluster to help ensure that the weight is not exceeded, resulting in suboptimal server performance.

### 3.4.2 Operational Considerations

Remote site gateways are monitored similarly to any other voice gateway. Voice availability is considered a critical service and, as such, requires an appropriate level of monitoring for both inbound and outbound PSTN connectivity.

### 3.4.3 Component Dependencies

The Cisco Unified Communications Manager Extension Mobility feature provides limited support in SRST mode (depending on the version Cisco IT has deployed). If clients are logged in from their home site, they will be able to make and receive calls from the PSTN if the DID extension on the phone matches the DID range terminating on one of the SRST routers. If the DID range does not terminate on the SRST router (for example, clients are logged in from a site other than their own or from a logged-off phone), only outbound calling to the PSTN is possible for that device.

H.323 IPVC endpoints will not remain survivable during loss of connectivity to the central cluster because they are not Cisco Unified Communications Manager controlled. No support exists within SRST for these devices.

Cisco Unified Video Advantage (formerly Cisco VT Advantage) also depends on the centralized Cisco Unified Communications Manager service and, as such, is not available in SRST mode.

In the current version supported by Cisco IT, Cisco IP Communicator endpoints and the Cisco Unified IP Phone 7920 are not supported in SRST mode. Due to the mobile nature of these devices, it is not possible to dynamically adopt calling patterns of a site other than that of the client’s home site. With this restriction, there is no guarantee that emergency calling will be routed to the correct destination while clients travel outside their home site. Therefore, a disclaimer explaining the limitations should be signed by the client using this device.

FXS, FXO, PRI, and BRI (both H.323 and MGCP) gateways terminating on SRST routers will remain available to same-site clients in SRST mode. All gateways (remote FXS, FXO, PRI, BRI, and ICT) remote to the site entering SRST mode will be unavailable until WAN services are restored.

Only local “remote site” conferencing resources are available to clients in SRST, and MoH will stream locally from the SRST router rather than from a central hub location.

If SRST capability is necessary across both WAN gateways to increase the amount of supported endpoints, Cisco IT recommends the following guidelines. To help ensure that all endpoints remain contactable in SRST conditions, it is necessary to configure dial-peers between the two routers. The dial-peers facilitate cross router calls between IP phones and inbound and outbound PSTN connectivity. If PSTN connections are distributed on a router other than that which an endpoint is registered to in SRST, the inter-router dial-peers help ensure that the endpoint remains reachable to and from the PSTN cloud. Because Cisco IT has the Extension Mobility feature enabled widely across theaters, a generic destination pattern dial-peer configuration is required to achieve this reach.

Example: VoIP dial-peers required for this configuration
Router A

- VoIP dial-peer 1, with destination 84081XXX and preference 1 points to CCM1
- VoIP dial-peer 2, with destination 84081XXX and preference 2 points to CCM2
- VoIP dial-peer 3, with destination 84081XXX and preference 3 points to Router B
  Optional if trunks for outbound PSTN calls terminate directly on Router A
- POTS dial-peer 4, with destination 9T and preference 1 – points to locally connected PSTN trunk
  Optional if trunks for overflow outbound PSTN calls terminate on Router B
- VoIP Dial-peer 5, with destination 9T and preference 2 – points to Router B

Router B

- VoIP dial-peer 1, with destination 84081XXX and preference 1 points to CCM1
- VoIP dial-peer 2, with destination 84081XXX and preference 2 points to CCM2
- VoIP dial-peer 3, with destination 84081XXX and preference 3 points to Router A
  Optional if trunks for outbound PSTN calls terminate directly on Router B
- Pots Dial-peer 4, with destination 9T and preference 1 – points to locally connected PSTN trunk
  Optional if trunks for overflow outbound PSTN calls terminate on Router A
- VoIP Dial-peer 5, with destination 9T and preference 2 – points to Router A

While in SRST mode, if a call is received on Router B and the destination IP phone is homed to Router A, the H.225 timer defined in Router B will expire for CCM1 and CCM2 (the WAN is down), and the call will be routed to Router A. The same applies for a call initiated in Router A for an IP phone hosted in Router B. The **alias** command is not configured as part of the SRST configuration. This configuration must not be configured using an IP-IP gateway between the two routers because a call initiated to a directory number, undefined in either, would loop due to the generic nature of the dial-peers.

## 3.4.4 Legal Considerations

When defining global standards for a corporate voice infrastructure, it is important to heed the regulatory restrictions and laws in the different countries affected by the design. The main areas of concern are the legality of VoIP and handling of emergency (eg., 911) calls.

### 3.4.4.1 VoIP Legality

Before implementing a remote site, it is necessary to determine the legal ramifications of enabling VoIP in the country of deployment. These laws must be investigated thoroughly. Cisco IT works closely with the Cisco legal department to help ensure that the technology implementations strictly adhere to the rules and regulations.

### 3.4.4.2 Emergency Calls

**Specific global guidelines (exception U.S. sites)**

Emergency Service Regulations (911 style service) should be in accordance with the local, state, and/or country regulations. Emergency PSTN connectivity (for the local Public Safety Answering Point, or PSAP) must be provided to remote site IP phones while in both normal operation and SRST mode.

**Specific U.S. guidelines**

For a remote site to comply with US regulation, Cisco IT works to ensure that the same service is available during normal Cisco Unified Communications Manager and SRST modes. To provide a valid DID callback number for each IP phone during both modes, a DID service is deployed and assigned to each IP phone. To help ensure connectivity to the PSTN in SRST mode, a local PSTN connection is necessary on at least one of the SRST routers. In this situation, the other SRST routers would have 9911 dial-peers pointing to the router with the PSTN connectivity, where the leading 9 denotes the PSTN access code. Additionally, coordinated testing with the PSAP is mandatory to help ensure that the service is functioning properly.

**All sites (no exceptions)**

Due to client mobility, it is vital to help ensure that emergency calls are based on the geographical location of the client at the time that the call is made. This procedure helps ensure that emergency services requested are local to that of the actual emergency. Hence, the emergency partition (where the emergency route patterns reside) is assigned to the phone's device profile to help ensure that the local emergency number is chosen regardless of the client's location. With mobile devices, this type of location-aware routing is not yet possible, so clients requesting these devices must sign a disclaimer acknowledging that they understand the emergency calling limitations of the device.

# 4 References

Cisco on Cisco Best Practice: How Cisco IT Standardizes Network Designs for Remote Locations
http://www.cisco.com/web/about/ciscoitatwork/network_systems/branch_office_network_design.html

Cisco on Cisco Case Study: How Cisco IT Migrated to Centralized Call Processing
http://www.cisco.com/web/about/ciscoitatwork/unified_comm/centralized_call_processing.html

Cisco on Cisco Best Practice: How Cisco IT Migrated to Enterprise-wide IP Telephony
http://www.cisco.com/web/about/ciscoitatwork/unified_comm/enterprise_ip_telephony_migration.html

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