



Cisco SIP SRST Integration with Cisco Small Business Pro SIP Phones using a 3rd-party Primary Proxy

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Revision History

Revision	Date	Author	Comment
1.2	08/27/09	Hanlin Fang, Nelson Seto, & Patrick Born	Initial version

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Overview

Cisco Unified SIP SRST provides a backup to an external SIP proxy server by providing basic registrar and back-to-back user agent (B2BUA) services. These services are used by SIP IP phones in the event of a network connection outage causing a SIP phone to lose contact with its primary SIP proxy server.

Cisco Unified SIP SRST enables SIP IP phones to continue to make and receive calls to and from the PSTN including off net calls to 911 services, and also to make and receive calls to and from other SIP IP phones. Full IOS LAN/WAN and packet voice features are supported on the SIP SRST router. However, not all features supported on the primary SIP proxy server are

available during SIP phone failover onto the Cisco SIP SRST. Cisco SIP SRST is currently [as of August 27, 2009] supported on the following Cisco routers:

- Cisco 880 Series
- Cisco 1861 Series
- Cisco 2800 Series
- Cisco 3800 Series Integrated Service Routers

This application note provides step-by-step instructions to enable SRST service for Cisco SMB Pro SIP phones with a 3rd-party primary SIP proxy.

The following topics are addressed in this document:

- The difference on the SRST failover process between Cisco and Cisco SMB Pro SIP phones.
- The features supported with Cisco SMB Pro SIP phones on Cisco SIP SRST
- The features not supported with Cisco SMB Pro SIP phones on Cisco SIP SRST
- The sample configuration to enable the support of Cisco SMB Pro SIP phones
- Troubleshooting tips on the SIP SRST

These topics are outside of the scope of this documentation:

- The primary SIP proxy setup
- The Cisco SMB Pro SIP phone provisioning procedures, refer to the Cisco SMB Pro IP phone menu for details

Topology

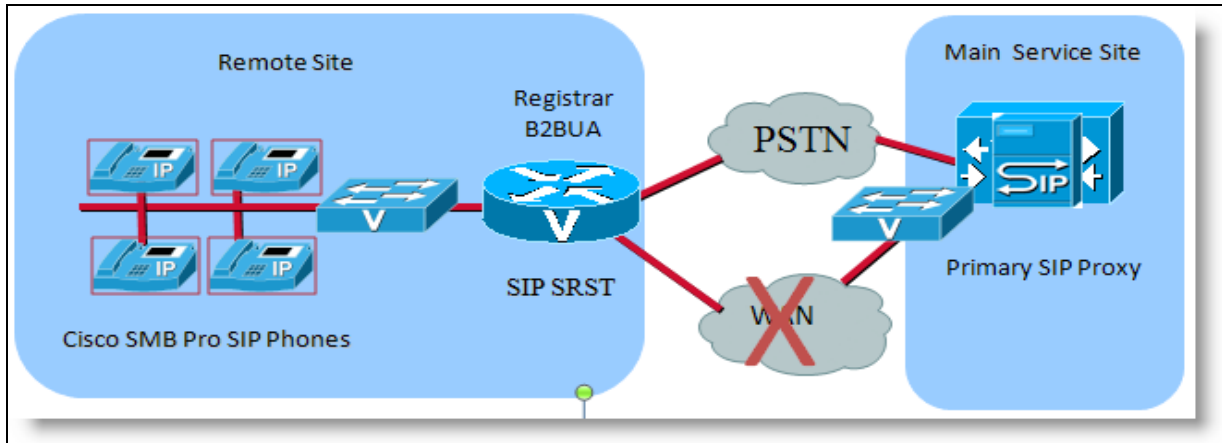


Figure 1-1: Network Topology for CUCM and SIP SRST

The CUCM and SIP SRST figure shows:

- The Primary SIP Proxy: SPA 9000 with the software version 6.1.5
- SIP SRST version: 4.1 (IOS 124-15.T)
- Cisco SMB Pro IP SIP phones:
 - SPA505x with the software version 7.1.3
 - SPA525G with the software version 7.2.5
 - SRST version 4.1

Only SIP back-to-back user agent (B2BUA) is configured. The older version of SIP SRST with Redirect server (SIP SRST 3.0) is not part of scope. The minimal SIP-SRST release to support SMB Pro SIP phones SRST 4.1.

Only UDP as the transportation protocol is tested in this document. However SIP SRST supports both TCP and UDP.

Differences between Cisco SIP phones and Cisco SMB Pro SIP phones During the SIP SRST Failover process

Cisco SIP phones will get the SRST address SRST reference parameter configured on CUCM and save it in the phones' TFTP configuration file. When a Cisco SIP IP Phone registers with the CUCM, a Cisco-proprietary keep-alive connection is established between the SIP phone and the Cisco SIP SRST. When the primary CUCM becomes unreachable, the SIP phone will switch over to the Cisco SRST. Meantime, the Cisco SIP phone starts polling the primary CUCM periodically in order to fall back when it becomes available.

There is no keep-alive communication between the Cisco SMB Pro SIP phones and the Cisco SRST service. The Cisco SRST service relies on the Cisco SMB Pro SIP phone's provisioning configuration to either have the explicit knowledge of the SRST address, or a connection to an address resolution service such as a DNS server in the network. This provides the SRST address to the SMB Pro SIP phone when the primary proxy is unreachable. Cisco SRST remains in a passive mode until the SMB SIP phones attempt to register to it.

Note:

In this application note, the SRST address is resolved by a DNS SRV resource record. The DNS server sets up the same DNS record for both primary SIP Proxy and Cisco SRST, but with

different weights. Cisco SMB Pro IP phones will reach the primary proxy or Cisco SRST based on the IP address returned by the DNS SRV server.

Features supported on Cisco SIP SRST with Cisco SMB Pro SIP phones

Cisco SRST supports Cisco SMB Pro SIP phones in the service provider's environment instead of the traditional enterprise UC solution in which Cisco UCM is the primary proxy for the SIP phones.

The Cisco SRST supports the following Cisco SMB Pro SIP phones:

- Cisco SPA525G
- Cisco SPA50x family

Cisco SRST supports Cisco SMB Pro SIP phones based on the RFC 3261 standard. Proprietary features are not supported. Cisco SIP SRST provides basic call control services during the failover situation. SRST supports the following features on Cisco SMB Pro SIP phones:

- Registration (both UDP and TCP)
- Basic call
- Ad-hoc 3-way conference
- Blind transfer
- Call forward
- Call hold/resume
- Caller ID Display
- DnD
- PSTN dial-out including 911
- PSTN incoming calls

Features not supported on Cisco SRST with Cisco SMB Pro SIP phones

Features not supported are:

- Time based call blocking
- Consult Transfer
- Distinctive ring
- Out-of-Dialog REFER
- Enhanced 911
- KPML
- System Message Display on the phone screen

Cisco SIP SRST Network Setup with Cisco SMB Pro SIP phones

This section addresses configuring the Cisco SMB Pro IP phones and the SRST router.

Configuring the Cisco SMB Pro SIP phone:

The Cisco SMB Pro SIP phones use DNS SRV resource records defined in RFC2782 in order to identify the primary and secondary proxies. In the event that you do not control the DNS servers, you can populate the Proxy field with the appropriate proxy SRV information. Both methods are demonstrated in this document.

- 1) Choose the SIP transport protocol and the port used for the Cisco SMB Pro SIP phone. (In this case, only UDP is configured)
Phone's web-ui > Voice tab > Ext N tab > SIP Settings:

SIP Settings			
SIP Transport:	UDP	SIP Port:	5060
SIP 100REL Enable:	UDP	EXT SIP Port:	
Auth Resync-Reboot:	TCP	SIP Proxy-Require:	
	TLS	Referor Bye Delay:	4
SIP Remote-Party-ID:	no	Referee Bye Delay:	0
Refer-To Target Contact:	no	Refer Target Bye Delay:	0
SIP Debug Option:	none	Auth INVITE:	no
Sticky 183:	no	Use Anonymous With RPID:	yes
Ntfy Refer On 1xx-To-Inv:	yes		

Note:

- Cisco SRST supports both UDP and TCP for transport protocols.
- TLS is not supported.
- Port 5060 is the default port used by Cisco SRST SIP transportation

- 2) Setup the SRST address resolution method on the Cisco SMB Pro SIP phone
 - **Configure the Proxy field as follows if you control the DNS servers:**
Phone's web-ui > Voice tab > Ext N tab > Proxy and Registration:

Proxy and Registration			
Proxy:	px.sip.switchspace.com		
Outbound Proxy:			
Use Outbound Proxy:	no	Use OB Proxy In Dialog:	no
Register:	yes	Make Call Without Reg:	no
Register Expires:	3600	Ans Call Without Reg:	no
Use DNS SRV:	yes	DNS SRV Auto Prefix:	no
Proxy Fallback Intvl:	3600	Proxy Redundancy Method:	Normal

Note:

px.sip.switchspace.com is the DNS record for both the primary SIP proxy and Cisco SIP SRST.

- **Configure the Proxy field as follows if you do not control the DNS servers:**
Phone's web-ui > Voice tab > Ext N tab > Proxy and Registration:



- MY_proxies:SRV=192.168.2.100:5060:p=1:w=50|192.168.2.22:5060:p=2:w=50**

Where:

- **My_proxies**—is an arbitrary name
 - **SRV=**—Implies a manual DNS SRV entry
 - **192.168.2.100:5060**—Defines the primary proxy and SIP registration port
 - **p=1**—Defines priority of the proxy, smaller numbers have higher priority
 - **w=50**—Defines the weight of the proxy, bigger numbers increase probability of being used. Refer to RFC2782 for additional information.
 - **|**—Separator character
 - **192.168.2.22:5060**—Defines the secondary proxy and SIP registration port
 - **p=2**—Defines this as the secondary proxy
- Use DNS SRV: yes
 - Proxy Fallback Intvl: default is 3600

Configuring SIP SRST:

The SIP Registrar on the SRST router accepts SIP Register requests and dynamically builds VoIP dial peers for each directory number assigned to the SIP phones. The dial-peers allow the SIP SRST router to route calls to SIP phones during the SRST mode. After SIP phones re-home back to the primary SIP proxy, the dial-peers will be erased from the SRST's database.

- 1) Enable SIP B2BUA and SIP Registrar Functionality on SIP SRST router

```
voice service voip
allow-connections sip to sip >>> see Note below for B2BUA capability enable
sip
registrar server max 600 min 60 >>> See Note below for SIP SRST
localhost dns:sipconnect-fca.atl0.cbeyond.net
```

Note:

- VoIP-to-VoIP connections permit termination and reorigination of transferred and forwarded calls over the VoIP network. For SIP SRST we enable SIP-to-SIP connections for hairpin call routing. The B2BUA that routes the call uses the SIP-to-SIP connection. The SIP B2BUA allows local SIP-SIP calls & SIP SRST supplementary services.
- The SIP registrar functionality in Cisco IOS software is a required part of Cisco SIP SRST to accept SIP phone registration
- Because the SIP registrar doesn't keep the persistent state information of registration, SIP endpoints need to send out registration message periodically. If the SIP SRST router is reloaded without proper timer configured, SIP SRST won't have knowledge of endpoints that registered to it before its reload

2) Enter “voice Register global” mode to allow globally assign characteristics to SIP phones.

```
voice register global
max-dn 200 >>> Max number of directory number on the SRST system, platform dependent
max-pool 100 >>> Max number of Pools on the SRST system, platform dependent
```

3) Configure the backup registrar (SIP SRST) services for SIP phones

```
voice register pool 1
id network 192.168.2.0 mask 255.255.255.0 >>> See below Note #1
call-forward b2bua busy 1525 >>> See below Note #2
call-forward b2bua noan 1525 timeout 10 >>> See below Note #2
```

Note:

- The **id** command identifies the individual SIP IP phone (You can use IP/MAC address to specify an IP phone) or a set of SIP IP phones (use network address to a group of IP phones) that are to be configured. This is required and must be configured before any other voice register pool commands.

4) To disable “REFER” based supplementary services

```
voice service voip
no supplementary-service sip moved-temporarily
no supplementary-service sip refer >>> To disable the SS using REFER
```

Note:

- With disabled SS configuration, REFER messages used for call transfer and redirect responses for call forwarding will be disabled from being sent to the destination by SRST. Hairpin will be used instead. This is the recommended configuration for SIP supplementary services.

5) Verify the SIP IP phone failover on SRST

```
show sip-ua status registrar
show voice register pool xx
show voice register dial-peers
show voice register statistics
```

6) Debug the SIP phone registration process

```
debug voice register errors
debug voice register events
debug ccsip message
```


Verifying IP Phone Fallback from the Primary SIP Proxy to the Local SRST Router

This session will explore the process between the Cisco SMB Pro SIP phone and Cisco SIP SRST with either the `show` CLI output or debug messages from the SRST router.

- 1) When a SIP IP Phone in the normal operation with the primary SIP proxy registration, "`show sip-ua status registrar`" on SRST router should return empty output.
- 2) "`show voice register dial-peer`" on SRST router should return nothing because no SIP phone has been registered with SRST yet
- 3) Now the WAN link is down, with the `debug ccsip` message on the SIP SRST router, the output should be

```
Received:
REGISTER sip:svctest.cisco.com SIP/2.0
Via: SIP/2.0/UDP 192.168.2.233:5062;branch=z9hG4bK-2fd1ae88
From: "1525" <sip:1525@svctest.cisco.com>;tag=6fec03ece9812173o2
To: "1525" <sip:1525@svctest.cisco.com>
Call-ID: 10c1ed35-efea0a6@192.168.2.233
CSeq: 39967 REGISTER
Max-Forwards: 70
Contact: "1525" <sip:1525@192.168.2.233:5062>;expires=50;+sip.instance="<00000000-0000-0000-0000-000E08DA0014>"
User-Agent: Cisco/SPA525G-7.2.5
P-Station-Name: spa525 ;mac=000e08da0014
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER, SUBSCRIBE
Allow-Events: dialog
Supported: replaces
```

```
Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.2.233:5062;branch=z9hG4bK-52584356
From: "1525" <sip:1525@svctest.cisco.com>;tag=6fec03ece9812173o2
To: "1525" <sip:1525@svctest.cisco.com>;tag=26228C-1969
Date: Fri, 20 Feb 2009 01:33:33 GMT
Call-ID: 10c1ed35-efea0a6@192.168.2.233
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 39968 REGISTER
Contact: <sip:1525@192.168.2.233:5062>;expires=60;x-cisco-newreg
Expires: 60
Content-Length: 0
```

Note:

The above capture is not a complete debug output. It only shows the relevant messages

- 4) Use the CLI "`show sip-ua status registrar`" command on the Cisco SIP SRST router. The line numbers with IP address should be shown.

```

2801-desk#show sip-ua status registrar
Line      destination  expires(sec) contact
          call-id
          peer
=====
4525      192.168.2.168  11      192.168.2.168
          5baf57c9-600173c@192.168.2.168
          40001
2525      192.168.2.234  43      192.168.2.234
          bd107bbc-1dba8be4@192.168.2.234
          40003
=====

```

Note:

- In the above case, 2 Cisco SMB Pro SIP phones with extensions 4525 and 2525 are registered with the Cisco SIP SRST.

5) Use the CLI “show voice register dial-peer” command on the SRST router

```

2801-desk# show voice register dial-peers
dial-peer voice 40001 voip
destination-pattern 4525
redirect ip2ip
session target ipv4:192.168.2.168:5060
session protocol sipv2
digit collect kpml
call-fwd-busy      1525
call-fwd-noan-timeou 5
call-fwd-noan      1525
after-hours-exempt FALSE

dial-peer voice 40003 voip
destination-pattern 2525
redirect ip2ip
session target ipv4:192.168.2.234:5061
session protocol sipv2
digit collect kpml
call-fwd-busy      1525
call-fwd-noan-timeou 5
call-fwd-noan      1525
after-hours-exempt FALSE

```

- 6) When the Cisco SMB Pro SIP IP phones re-home to the primary SIP proxy, they first un-register from the Cisco SIP SRST router and then send a normal SIP REGISTER message to the primary SIP proxy.

Sample SIP SRST Configuration

Building configuration...

Current configuration : 7015 bytes

```
!  
version 12.4  
service timestamps debug datetime msec  
service timestamps log datetime msec  
no service password-encryption  
!  
hostname 2801-desk  
!  
boot-start-marker  
boot system flash flash:c2801-adventerprisek9_ivs-mz.124-15.T.bin  
boot-end-marker  
!  
logging buffered 2000000  
logging console informational  
enable password cisco  
!  
no aaa new-model  
ip cef  
!  
no ip dhcp use vrf connected  
!  
ip dhcp pool test  
!  
ip name-server 171.70.168.183  
ip dhcp-client update dns  
!  
multilink bundle-name authenticated  
!  
voice-card 0  
!  
voice service voip  
  allow-connections sip to h323  
  allow-connections sip to sip  
  no supplementary-service h450.2  
  no supplementary-service h450.3  
  no supplementary-service sip moved-temporarily  
  no supplementary-service sip refer  
  h323  
  sip  
    registrar server expires max 600 min 60  
    localhost dns:sipconnect-fca.atl0.cbeyond.net  
    no update-callerid  
    no call service stop  
!  
!  
voice register global  
  max-dn 12  
  max-pool 12  
!
```

```

voice register pool 1
id network 192.168.2.0 mask 255.255.255.0
call-forward b2bua busy 1525
call-forward b2bua noan 1525 timeout 5
!
voice translation-rule 9
rule 1 /^911$/ /911/
rule 2 /^9(.*)/ /1/
!
voice translation-rule 10
rule 1 /^.*/ /6783979412/
!
voice translation-rule 100
rule 1 /^1519(.*)/ /519\1/
rule 2 /^226(.*)/ /226\1/
rule 3 /^1(.*)/ /1\1/
rule 4 /^[2-9].*/ /011\1/
!
voice translation-profile OUT
translate called 1
!
voice translation-profile PSTN_Outgoing
translate calling 10
translate called 9
!
voice translation-profile strip+
translate called 100
!
voice translation-profile strip1
translate called 1
!
application
service EA flash:app-cme-ea-2.0.0.0.tcl
param space english index 0
param space english language en
param space english location flash:
param ea-password 1234
param space english prefix en
!
global
service alternate default
!
crypto pki trustpoint TP-self-signed-1941349233
enrollment selfsigned
subject-name cn=IOS-Self-Signed-Certificate-1941349233
revocation-check none
rsa keypair TP-self-signed-1941349233
!
crypto pki certificate chain TP-self-signed-1941349233
certificate self-signed 01
3082024C 308201B5 A0030201 02020101 300D0609 2A864886 F70D0101 04050030
31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
69666963 6174652D 31393431 33343932 3333301E 170D3039 30323230 30303532
35315A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D31 39343133
34393233 3330819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281

```

```
8100ADCA 5B9B230B 385FD9BB DB7A12B2 4D7D32A3 0267011E 53574529 BA09DE4E
4A50252C 84A0B743 4E1AE7AD F4A78595 90C9DC09 5F855C88 B45DBA22 B055E5DA
D0D6C3A1 1F9430FE C26DB829 AC82BDF4 FFA86B53 18639C4C B2D236FF B2676818
037789E6 92BBBBA4 BE8A6616 9F207443 172D250F 012EAF79 562EB67C 786655F9
49930203 010001A3 74307230 0F060355 1D130101 FF040530 030101FF 301F0603
551D1104 18301682 14323830 312D6465 736B2E73 69707572 612E636F 6D301F06
03551D23 04183016 80146D88 E6DC5008 7D77DB5F 9BFF94D1 666AEEA5 EC85301D
0603551D 0E041604 146D88E6 DC50087D 77DB5F9B FF94D166 6AEEA5EC 85300D06
092A8648 86F70D01 01040500 03818100 956B16CF D4EBF599 AFA51BB5 03305034
3F7439EF B4A42001 19664164 62C739C9 53DF6484 E961F879 1DC566B8 AE96427E
B55288F4 534A5E21 CAF95C33 70E9439D 9060729F 2D29353E 4FBAE360 39423A38
E99CFB82 20409AAC 4147353F 579DF49C 2758280F 27C7D2B2 F2B0FF89 FC00A676
FB16782E 18A85E29 2C85C1BE AAAE454A
```

```
quit
!
archive
log config
hidekeys
!
ip ftp username cisco
ip ftp password cisco
!
interface FastEthernet0/0
description GRE to WAN
ip dhcp client hostname nseto-c2800
ip address dhcp
ip nat outside
ip virtual-reassembly
duplex auto
speed auto
!
interface FastEthernet0/1
description to CME-POD1
ip address 1.1.100.254 255.255.255.0
ip nat inside
ip virtual-reassembly
shutdown
duplex auto
speed auto
!
ip route 0.0.0.0 0.0.0.0 128.107.148.1
ip route 192.168.10.0 255.255.255.0 1.1.100.1
!
ip http server
ip http secure-server
ip nat inside source list 105 interface FastEthernet0/0 overload
!
control-plane
!
voice-port 0/1/0
!
voice-port 0/1/1
!
voice-port 0/1/2
!
voice-port 0/1/3
```

```

!
dial-peer cor custom
  name pstn-in
  name pstn-int
!
dial-peer cor list pstn-int
  member pstn-int
!
dial-peer cor list pstn
  member pstn-in
  member pstn-int
!
dial-peer voice 100 voip
  translation-profile outgoing PSTN_Outgoing
  destination-pattern 1[2-9]..[2-9].....
  voice-class sip dtmf-relay force rtp-nte
  session protocol sipv2
  session target sip-server
  dtmf-relay rtp-nte
  codec g711ulaw
  ip qos dscp cs5 media
  ip qos dscp cs4 signaling
  no vad
!
dial-peer voice 54 pots
  description outgoing call-local
  preference 5
  destination-pattern 9[2-9].....
  port 0/1/0
  forward-digits 7
  no sip-register
!
dial-peer voice 55 pots
  description outgoing call-domestic
  preference 5
  destination-pattern 91[2-9]..[2-9].....
  prefix 1
  no sip-register
!
sip-ua
  authentication username 6783979412 password 13524F405C5457
  no remote-party-id
  retry invite 2
  retry register 10
  retry options 1
  timers connect 100
  registrar dns:sipconnect-fca.atl0.cbeyond.net expires 3600
  sip-server dns:sipconnect-fca.atl0.cbeyond.net
  host-registrar
  g729-annexb override
!
gatekeeper
  shutdown
!
line con 0
line aux 0

```

```
line vty 0 4
exec-timeout 35791 0
privilege level 15
no login
transport input telnet ssh
line vty 5 15
privilege level 15
login local
transport input telnet ssh
!
scheduler allocate 20000 1000
!
webvpn cef
event manager applet test
event none
action 0.1 cli command "config t"
action 0.2 cli command "host test"
action 0.3 cli command "end"
action 1.0 cli command "telnet 10.100.101.10"
action 1.1 cli command "config t" pattern "SIPConnectCME#"
action 1.2 cli command "host SIP" pattern "SIPConnectCME(config)#"
action 1.4 cli command "end"
action 1.5 cli command "quit"
action 1.6 syslog msg "CONFIG DONE"
!
End
```

References:

- The SIP SRST Administrator Guide on http://www.cisco.com/en/US/docs/voice_ip_comm/cusrst/admin/sipsrst/configuration/guide/spsrst0.html
- SIP SRST implementation Guide on SRST wiki page <http://zed.cisco.com/confluence/display/VOICE/SRST>
- RFC2782: A DNS RR for specifying the location of services (DNS SRV): <http://tools.ietf.org/html/rfc2782>
- [Cisco Small Business Pro IP Phone Administration Guide](http://www.cisco.com/en/US/docs/voice_ip_comm/csbpipp/ip_phones/administration/guide/spa500_admin.pdf) on: http://www.cisco.com/en/US/docs/voice_ip_comm/csbpipp/ip_phones/administration/guide/spa500_admin.pdf




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