AASTRA®

Clearspan®

CISCO 7940 & 7960
SIP Phone Configuration

2813-001
Release 14.0

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Revision History

The following represents the revision history of this publication:

Revision Number	Date Completed	Point of Contact	Description
001	01/15/09	Bev Marsh	Initial release of this publication.

Table of Contents

1 Overview

This document describes the configuration procedures required for a Cisco 7940/7960 SIP Phone to make full use of the capabilities of Clearspan.

The 7940/7960 is one of the many access devices that interoperate with Clearspan.

The 7940/7960 uses the Session Initiation Protocol (SIP) to communicate with Clearspan for call control. It also translates voice to audio packets for transmission across a packet network.

This guide describes the specific configuration items that are important for use with Clearspan. It does not describe the purpose and use of all configuration items on the 7940/7960. For those details, see the Cisco SIP IP Phone Administrator Guide, provided by Cisco Systems, Inc.

2 Clearspan Validation Package Support Level

Devices are validated according to Clearspan Validation Packages. Each package validates a subset of features or items. This section describes the device's support level for a Clearspan Validation Package as well as the features or items in the package that are not supported. For specific issues, see section 3.2 Interoperability Issues. For a complete list of items validated per package, see Appendix A: Clearspan Validation Package Test Items.

Clearspan Package	Support Level	Items Not Supported
Basic Call	Full	
Clearspan Enhanced Services	Partial	Advanced Call Control CommPilot Call Manager
DUT Services – Call Control	Partial	Network Conference Three-Way Network Conference N-Way
DUT Services – Registration and Authentication	Partial	Re-registration after Registration Rejection
DUT Services – Fax	None	
DUT Services – Busy Lamp Field	None	
Redundancy	Full	
SBC/ALG	Full	
Shared Call Appearance	None	
Feature Key Synchronization	None	
Video	None	
TCP	None	

3 Device Capabilities and Known Interoperability Issues

This section describes the features supported by the Cisco 7940/7960 SIP Phone, as well as Clearspan interoperability issues and impact. The following table describes capabilities.

Verified Revisions shows the results of testing a specific Clearspan version with a specific partner's device under test (DUT) version.

Compatible Revisions indicates the maintenance versions that should interface properly with Clearspan.

NOTE 1: Aastra validates that the device works properly with the Clearspan SIP interface. Aastra does not validate qualitative aspects of the device or other device capabilities, which are outside the scope of the SIP signaling interface. For device feature and performance testing results, consult Cisco.

NOTE 2: Aastra generally tests only the latest generally available (GA) device firmware/software with latest GA Clearspan release. If there is a need to use a non-validated mix of Clearspan and device software versions, customers can mitigate their risk by self-testing the combination using the appropriate Clearspan Release Test Plan.

3.1 Capabilities

Device Type	SIP Phone
Lines and Appearances	7960: 6 lines, with 2 appearances per line 7940: 2 lines, with 2 appearances per line
Speaker/Power/Bridge	Yes/Yes/Yes
Verified Revisions	Clearspan Release 14.sp1: P0S3-08-7-00
Compatible Revisions	Any maintenance release of the verified revision.
SIP Proxy FQDN DNS Lookup (A, SRV, NAPTR)	A, SRV
Outbound Proxy Configurable	Yes
Outbound Proxy FQDN DNS Lookup (A, SRV, NAPTR)	A, SRV
Clearspan Redundancy Enabled	Yes
Clearspan Shared Call Appearance	Not Applicable
Clearspan Enhanced IP Phone Configuration	Yes

Device Services	• CLID Blocking • Anonymous Call Rejection • Call Forward Always • Do Not Disturb • Remote Restart • Call Hold • Call Conference • Attended Call Transfer • Blind Call Transfer
Device Call Control (Device-Controlled or Flash INFO-based)	Device-Controlled
Codecs	G.711u, G.711a, G.729a
RFC 2833 DTMF	Yes
T.38 Fax	Not Applicable
TCP	No
TLS	No

3.2 Interoperability Issues

This section lists the known interoperability issues between Clearspan and partner release(s). For more information on issues related to the particular software release, see the partner release notes.

ExtraView Issue	Issue Title and Description	Partner Releases			
		P0S3-07-4-00	P0S3-07-5-00	P0S3-08-7-00	
22666	**Redundancy: 7960 does not properly drop the first call in failover situation** The 7960 does not properly tear down a call established on the primary Application Server when it fails over to the secondary Application Server to answer a second call, and receives a 481 when it tries to hold the first call. Workaround: None.	X	X	X	
22649	**One way voice when hold from CM, then Hold/Retrieve from phone** If the call is held from the Call Manager, and then held and retrieved from the phone, the 7960 does not stream media to the remote end. It still thinks it is remotely held. Workaround: Perform another hold/retrieve from the Call Manager.	X			

ExtraView Issue	Issue Title and Description	Partner Releases			
17174	**A rejected registration (403 Forbidden) causes a re-attempt every 60 seconds** When Clearspan returns a 403 Forbidden response to a registration, the 7960 should retry the registration every 10 to 20 minutes. Workaround: None.	X	X	X	

4 Clearspan Device Identity/Profile

Clearspan configurable device identify/profile is introduced in Clearspan Release 14.0. **This section applies only to Clearspan Release 14.0 and later**.

The following table identifies the required Clearspan device identity/profile settings for interoperability between the 7940/7960 and Clearspan. For an explanation of the profile parameters, refer to the Clearspan Device Inventory Guide.

4.1 Cisco 7940/7960 Identify/Device Profile

Signaling Address Type	Intelligent Proxy Addressing
Number of Lines	7940: 2 7960: 6
Ringback Tone/Early Media Support	Local Ringback – No Early Media
Authentication	Disabled
Registration Capable	X
Static Registration Capable	
E.164 Capable	
Trusted	
Authenticate REFER	
Authentication Override	
Video Capable	
RFC 3264 Hold	X
Route Advance	
Wireless Integration	
PBX Integration	
Use Business Trunking Contact	
Forwarding Override	
Conference Device	
Music On Hold Device	
Auto Configuration Soft Client	
Web Based Configuration URL	
Auto Configuration Type	2 Config File
Reset Event	checkSync
Enable Monitoring	
CPE System File Name	SIPDefault.cnf
Device File Format	SIP%BWMACADDRESSUPPER%.cnf

5 Configuration

The 7940/7960 can be configured with a configuration file using the Trivial File Transfer Protocol (TFTP). The following examples describe how to set the parameters using a configuration file. This configuration description assumes the 7940/7960 will use the Dynamic Host Configuration Protocol (DHCP) to obtain an IP address, TFTP server, and other network settings. The 7940/7960 should be configured to load the configuration file each time it resets or re-synchronizes. For detailed information on automated provisioning, see the Cisco SIP IP Phone Administrator Guide.

The capabilities of the 7940/7960 have been verified for use with Clearspan based on the settings described in the following table. For more information on the meaning, purpose, and applicability of the individual configuration items see the Cisco SIP IP Phone Administrator Guide.

5.1 Configuration Files

Files Provided by Partner	Level	Description
SIPDefault.cnf	System	Contains configurable parameters that apply to all devices in a given deployment.
dialplan.xml	System/Subscriber	The XML file for the dial plan specification.
SIP<mac-address>.cnf Example: SIP0003E3630C94.cnf	Subscriber	Contains configurable parameters that apply to an individual device in a deployment.

5.2 System Level Configuration

Step	Command	Purpose
System Level Configuration Parameters		

Step 1	**Enter the SIP proxy address.**
	Example:
	`proxy1_address: "as.mycompany.com"`
	`proxy1_port: 5060`
		Use the domain name of the AoR, provisioned in the Line/Port setting of the Clearspan subscriber's device configuration.
		The domain name should resolve to the Application Server cluster.
		Configure for each 7940/7960 line in use by different user.

Step 1 (alternate)	**Enter the Outbound Proxy address.**
	Example:
	`outbound_proxy: "alg.aastra.com" or "66.160.10.32"`
	`outbound_proxy_port: 5060`
		Set the Outbound Proxy to the VRRP IP address or FQDN of SIP Session Manager (SSM).

Step 2	**Disable privacy_03 support on the phone.** remote_party_id: 0	The 7940/7960 does not implement *draft-ietf-sip-privacy-03.txt* in a manner that interoperates with Clearspan, so this must be disabled on the phone. The default CLID blocking mechanism that the phone uses in its place interoperates with Clearspan.
Step 3	**Enable phone registration.** proxy_register: 1 timer_register_expires: 86400	The 7940/7960 must be configured to register with the Clearspan Application Server. Registration is necessary for Clearspan to terminate calls to the 7940/7960. The suggested registration period is one day.
Step 4	**Enter SIP timer/retry parameters.** timer_t1: 500 timer_t2: 4000 sip_invite_retx: 3 sip_retx: 3	These values allow the 7940/7960 to be tuned with respect to network and traffic delays and to allow for reasonable call setup times in failover situations.
Step 5	**Enable RFC 2833 (out-of-band) DTMF.** dtmf_outofband: **avt-avt** dtmf_inband: **1** dtmf_avt_payload: 101	Configure the 7940/7960 to enable DTMF mode negotiation (inband or out-of-band).
Step 6	**Set the preferred codec.** Example: preferred_codec: g711ulaw	The 7940/7960 supports G.711 µlaw, G.711 alaw, and G.729. This parameter specifies which codec is preferred.
Step 7	**Disable VAD to prevent voice clipping.** enable_vad: 0	VAD or silence suppression must be disabled when the 7940/7960 is deployed behind a NAT. The NAT binding does not open until the 7940/7960 streams media which does not occur with VAD enabled until the user speaks. This means that with VAD enabled, the caller on a 7940/7960 does not hear the callee until the caller speaks. Note that VAD is disabled by default on the 7940/7960.
Step 8 (Optional)	**Allow conference join on hang up.** cnf_join_enable: 1	Optional: Allow the bridge on a Three-Way Call to join remaining parties on hang-up.
Step 9 (Optional)	**Allow transfer while ringing.** semi_attended_transfer: 1	Optional: Allow the transfer to be completed while the target phone is still ringing.

5.3 Subscriber Level Configuration Parameters

This section identifies the device-specific parameters, including registration and authentication. These settings must be unique across devices to be matched with the settings for a Clearspan subscriber.

Provisioning a subscriber to register with Clearspan allows calls to terminate to the subscriber's line. Registration requires that a unique Address of Record (AoR) is provisioned on Clearspan and the phone; provisioning an AoR on Clearspan consists of setting the line/port parameter to a unique value within the Application Server cluster.

Step	Command	Purpose
Phone-specific Configuration File SIP<mac-address>.cnf		

Step 1	**`Enter the line name.`**
	`Example:`
	`line1_name: "2405551111"`
		The line name must correspond with the Line/Port setting of the Clearspan subscriber's device configuration.

Step 2	**`Enter the authentication parameters.`**
	`Example:`
	`line1_authname: "1111@as.mycompany.com"`
	`line1_password: "welcome"`
		If the Authentication service is configured on Clearspan, these parameters must be configured to match the Clearspan settings.

Step 3
(Optional)
	`Enter Message URI.`
	`Example:`
	`messages_uri: "2405559999"`
		Enter the Clearspan Voice Portal number or other Voice Messaging Server number.

5.4 Shared Call Appearance Configuration

The 7940/7960 does not support the Clearspan implementation of this feature.

6 Enhanced IP Phone Configuration

Enhanced IP Phone Configuration is a Clearspan feature that enables automatic generation of device configuration files, given administrator-supplied templates.

For more information on the Enhanced IP Phone Configuration feature, see the Enhanced IP Phone Configuration Guide.

For sample 7940/7960 system and group template files, see System-wide Parameter File SIPDefault.cnf.

The group template file is used to build the configuration files for the Cisco 7940/7960 IP Phones assigned to the group.

To use this feature to upgrade the firmware on the 7940/7960, the group template file should be modified to reference the new firmware version and then the phone(s) should be reset from the *Device Configuration* web page.

NOTE: Aastra does not manage or distribute template files for use with the Enhanced IP Phone Configuration feature. Obtain template files from Cisco or use the configuration files obtained from Cisco for the specific Cisco 7940/7960 SIP Phone firmware release to create template files appropriate for your installation.

7 Appendix A: Sample 7940/7960 Configuration Files

NOTE: The following samples are examples and should be used as a reference only. DO NOT CUT AND PASTE THESE EXAMPLES TO GENERATE YOUR CONFIGURATION FILES. Use the configuration files obtained from Cisco with the specific release to generate your configuration files.

This section includes samples of the following files:

- System-wide Parameter File SIPDefault.cnf
- Phone-specific File SIP<mac-address>.cnf
- Dial Plan Template File (dialplan.xml)
- Group Template File

7.1 System-wide Parameter File SIPDefault.cnf

NOTE: This is an example file and should be used for reference only.

```
# SIP Default Generic Configuration File

# Image Version
image_version: P0S3-08-7-00

# Proxy Server
proxy1_address: as.mycompany.com

# Proxy Server Port (default - 5060)
proxy1_port: 5060

# Proxy Registration (0-disable (default), 1-enable)
proxy_register: 1

# Phone Registration Expiration [1-3932100 sec] (Default - 3600)
timer_register_expires: 86400

# Codec for media stream (g711ulaw (default), g711alaw, g729)
preferred_codec: g711ulaw

# TOS bits in media stream [0-5] (Default - 5)
tos_media: 5

# Inband DTMF Settings (0-disable, 1-enable (default))
dtmf_inband: 1

# Out of band DTMF Settings (none-disable, avt-avt enable (default), avt_always
- always avt )
dtmf_outofband: avt

# DTMF dB Level Settings (1-6dB down, 2-3db down, 3-nominal (default), 4-3db
up, 5-6dB up)
dtmf_db_level: 3

# SIP Timers
timer_t1: 500 ; Default 500 msec
```

```
timer_t2: 4000 ; Default 4 sec
sip_retx: 3 ; Default 11
sip_invite_retx: 3 ; Default 7
timer_invite_expires: 180 ; Default 180 sec

####### New Phase 4 Parameters #######

# Dialplan template (.xml format file relative to the TFTP root directory)
dial_template: dialplan

# TFTP Phone Specific Configuration File Directory
tftp_cfg_dir: "" ; Example: ./sip_phone/

# Time Server (There are multiple values and configurations refer to Admin
Guide for Specifics)
sntp_server: 129.6.15.29 ; SNTP Server IP Address
sntp_mode: directedbroadcast ; unicast, multicast, anycast, or
directedbroadcast (default)
time_zone: EST ; Time Zone Phone is in
dst_offset: 1 ; Offset from Phone's time when DST is in effect
dst_start_month: April ; Month in which DST starts
dst_start_day: "" ; Day of month in which DST starts
dst_start_day_of_week: Sun ; Day of week in which DST starts
dst_start_week_of_month: 1 ; Week of month in which DST starts
dst_start_time: 02 ; Time of day in which DST starts
dst_stop_month: Oct ; Month in which DST stops
dst_stop_day: "" ; Day of month in which DST stops
dst_stop_day_of_week: Sunday ; Day of week in which DST stops
dst_stop_week_of_month: 8 ; Week of month in which DST stops 8=last week of
month
dst_stop_time: 2 ; Time of day in which DST stops
dst_auto_adjust: 1 ; Enable(1-Default)/Disable(0) DST automatic adjustment

# Instruct the phone not to use draft-ietf-sip-privacy-03.txt for
# calling line ID blocking (CIDB)
remote_party_id: 0 ; Use “Anonymous” to indicate CIDB

# Do Not Disturb Control (0-off, 1-on, 2-off with no user control, 3-on with no
user control)
dnd_control: 0 ; Default 0 (Do Not Disturb feature is off)

# Caller ID Blocking (0-disbaled, 1-enabled, 2-disabled no user control, 3-
enabled no user control)
callerid_blocking: 0 ; Default 0 (Disable sending all calls as anonymous)

# Anonymous Call Blocking (0-disbaled, 1-enabled, 2-disabled no user control,
3-enabled no user control)
anonymous_call_block: 0 ; Default 0 (Disable blocking of anonymous calls)

# DTMF AVT Payload (Dynamic payload range for AVT tones - 96-127)
dtmf_avt_payload: 101 ; Default 100

# Speed Dial Key for Message Key
messages_uri: 2405559999

#Enable or Disable VAD (0-disabled (default), 1-enabled)
enable_vad: 0 ; Disable VAD when deployed behind a NAT

# NAT/Firewall Traversal
nat_enable: 1 ; 0-Disabled (default), 1-Enabled
nat_address: "" ; WAN IP address of NAT box (dotted IP or DNS A record only)
voip_control_port: 5060 ; UDP port used for SIP messages (default - 5060)
start_media_port: 16384 ; Start RTP range for media (default - 16384)
```

```
end_media_port: 32766 ; End RTP range for media (default - 32766)
nat_received_processing: 0 ; 0-Disabled (default), 1-Enabled

# Outbound Proxy (SSM or EdgeMarc) Support
outbound_proxy: "alg.aastra.com" ;
outbound_proxy_port: 5060 ; default is 5060

####### New Parameter added in Release 3.0 #######
# Allow for the bridge on a 3way call to join remaining parties upon hang up
cnf_join_enable : 1 ; 0-Disabled, 1-Enabled (default)

####### New Parameters added in Release 3.1 #######
# Allow Transfer to be completed while target phone is still ringing
semi_attended_transfer: 1 ; 0-Disabled, 1-Enabled (default)

# Telnet Level (enable or disable the ability to telnet into the phone)
telnet_level: 1 ; 0-Disabled (default), 1-Enabled, 2-Privileged
phone_password: cisco
```

7.2 Phone-specific File SIP<mac-address>.cnf

NOTE: This is an example file and should be used for reference only.

```
################################################################################
#
############ Using the default device configuration file that was
############ provided with the system
################################################################################
#

#This is the configuration file is for device ID: Cisco-7960-003094c3ed66 Built
on: 2004.04.30 15:33:05:018 EDT

#Value that changes to cause the phone to reboot when a NOTIFY SIP message is
received sync: 2004.04.30 15:33:05: EDT


################### LINE 1 FOR Scooby 9313Doo ##########################
##############################################################################
#######
#
# Image Version
image_version: P0S3-08-7-00

# Line 1 Registration Authentication
line1_authname: "1111@as.mycompany.com"

# Line 1 Registration Password
line1_password: "welcome"

# Line 1 appearance
line1_name: "2405551111"

# Line 1 Display Name (Display name to use for SIP messaging)
line1_displayname: "Scooby Doo"

#Short Name:
Line1_shortname: "2405551111"
<<< Line 2-6 Configuration has been removed for space >>>>
```

```
# Phone Prompt (The prompt that will be displayed on console and telnet)
phone_prompt: "SIP Phone" ; Limited to 15 characters (Default - SIP Phone)

# Phone Password (Password to be used for console or telnet login)
phone_password: "123" ; Limited to 31 characters (Default - cisco)

# User classification used when Registering [ none(default), phone, ip ]
user_info: none

####### New Release 3 Parameters ######
#phone_password:
####### New Phase 4 Parameters #######

# Phone Label (Text desired to be displayed in upper right corner - Has no
effect on SIP messaging)
phone_label: ""
```

7.3 Dial Plan Template File (dialplan.xml)

NOTE: This is an example file and should be used for reference only.

```
<DIALTEMPLATE>
<TEMPLATE MATCH="0" Timeout="3" /> <!-- Local operator-->
<TEMPLATE MATCH="011*" Timeout="6" /> <!-- International calls-->
<TEMPLATE MATCH="00" Timeout="3" /> <!-- LD Operator-->
<TEMPLATE MATCH="0.........." Timeout="3" /> <!-- Operator assisted-->
<TEMPLATE MATCH=".11" Timeout="1" /> <!-- Service numbers -->
<TEMPLATE MATCH="101....1.........." Timeout="1" /> <!-- EqualAccess Service-->
<TEMPLATE MATCH="1.........." Timeout="0" /> <!-- Long Distance -->
<TEMPLATE MATCH="......." Timeout="3" /> <!-- 7 Digit Dialing -->
<TEMPLATE MATCH=".........." Timeout="3" /> <!-- 10 Digit Dialing -->
<TEMPLATE MATCH="9.........." Timeout="2" /> <!-- 9+dialing -->
<TEMPLATE MATCH="...." Timeout="3" /> <!-- Extension Dialing 4 Digits -->
<TEMPLATE MATCH="....." Timeout="3" /> <!-- Extension Dialing 5 Digits -->
<TEMPLATE MATCH="*" Timeout="15"/> <!-- Anything else -->
</DIALTEMPLATE>
```

7.4 Group Template File

NOTE: This is an example file and should be used for reference only.

```
# SIP Configuration Generic File
#This is the configuration file for device ID: %BWDEVICEID% Built on:
%BWTIMESTAMP%
#Value that changes to cause the phone to reboot when a NOTIFY SIP message is
received sync: %BWTIMESTAMP%

# Image Version
image_version: P0S3-07-4-00

##### LINE 1 FOR %BWFIRSTNAME-1% %BWLASTNAME-1% #####
#########################################################################
# Line 1 Registration Authentication
line1_authname: %BWAUTHUSER-1%

# Line 1 Registration Password
line1_password: %BWAUTHPASSWORD-1%
```

```
# Line 1 appearance
line1_name: %BWLINEPORT-1%

# Line 1 Display Name (Display name to use for SIP messaging)
line1_displayname: %BWCLID-1%

#####LINE 2 FOR %BWFIRSTNAME-2% %BWLASTNAME-2% #################
########################################################################
# Line 2 Registration Authentication
line2_authname: %BWAUTHUSER-2%

# Line 2 Registration Password
line2_password: %BWAUTHPASSWORD-2%

# Line 2 appearance
line2_name: %BWLINEPORT-2%

# Line 2 Display Name (Display name to use for SIP messaging)
line2_displayname: %BWCLID-2%

######### LINE 3 FOR %BWFIRSTNAME-3% %BWLASTNAME-3%
#######################################################################
# Line 3 Registration Authentication
line3_authname: %BWAUTHUSER-3%

# Line 3 Registration Password
line3_password: %BWAUTHPASSWORD-3%

# Line 3 appearance
line3_name: %BWLINEPORT-3%

# Line 3 Display Name (Display name to use for SIP messaging)
line3_displayname: %BWCLID-3%

################### LINE 4 FOR %BWFIRSTNAME-4% %BWLASTNAME-4%
#####################################################################
#
# Line 4 Registration Authentication
line4_authname: %BWAUTHUSER-4%

# Line 4 Registration Password
line4_password: %BWAUTHPASSWORD-4%

# Line 4 appearance
line4_name: %BWLINEPORT-4%

# Line 4 Display Name (Display name to use for SIP messaging)
line4_displayname: %BWCLID-4%

################### LINE 5 FOR %BWFIRSTNAME-5% %BWLASTNAME-5%
#######################################################################
#
# Line 5 Registration Authentication
line5_authname: %BWAUTHUSER-5%

# Line 5 Registration Password
line5_password: %BWAUTHPASSWORD-5%

# Line 5 appearance
line5_name: %BWLINEPORT-5%

# Line 5 Display Name (Display name to use for SIP messaging)
```

```
line5_displayname: %BWCLID-5%

################### LINE 6 FOR %BWFIRSTNAME-6% %BWLASTNAME-6%
#########################################################################
# Line 6 Registration Authentication
line6_authname: %BWAUTHUSER-6%

# Line 6 Registration Password
line6_password: %BWAUTHPASSWORD-6%

# Line 6 appearance
line6_name: %BWLINEPORT-6%

# Line 6 Display Name (Display name to use for SIP messaging)
line6_displayname: %BWCLID-6%

# Phone Label (Text desired to be displayed in upper right corner)
phone_label: %BWDISPLAYNAMELINEPORT% ; Has no effect on SIP messaging
```

8 Appendix B: Clearspan Validation Package Test Items

The following table describes the items tested in each Clearspan Validation Package.

Clearspan Validation Package	Items Supported
Basic Call	• Basic Call Origination/Termination • Call Failure Codes • Session Audit • Ringback • Dial Plan • Inband DTMF • RFC 2833/Negotiation • DTMF Relay • Codec Renegotiation
Clearspan Enhanced Services	• Basic CommPilot Call Manager Functions • Voice Messaging Audio MWI • Voice Messaging Visual MWI • Ring Splash • Priority Alerting • Priority Call Waiting • Alternate Numbers • Advanced Call Control • Anonymous Call • Remote Restart • Call Park Retrieve – Answer with Hold
DUT Services – Call Control	• Call Waiting • Call Hold • Blind Transfer • Attended Transfer • Three-Way Call • Network Conference Three-Way • Network Conference N-Way

Clearspan Validation Package	Items Supported
DUT Services – Registration and Authentication	• Authenticated Registration • Maximum Registration • Minimum Registration • Rejected Registration • Authenticated Origination • Authenticated Re-INVITE • Authenticated REFER • Clearspan Authentication
DUT Services – Fax	• Fax Passthrough • Fax T38
DUT Services – Busy Lamp Field	• Basic BLF
Redundancy	• DNS • Registration Failover • Call Setup Failover • Mid-Call Failover
SBC/ALG	• Registration • Call Origination • Call Termination
Shared Call Appearance	• Line Seize • Line Lamp Management • Line Hold/Retrieve • Multiple Call Arrangement • SCA Bridging
Feature Key Synchronization	• Feature Key Synchronization
Video	• Integrated Video Phone • Video Services • Video Add-On
TCP	• Basic

9 References

- *Cisco 7940/7960 SIP IP Phone Administrator Guide.* Cisco, March 2005, Cisco SIP IP Phone Administrator Guide, CiscoSIPPhoneAdminGuide.pdf. Available from http://www.cisco.com.
- 2750 Clearspan Device Inventory Guide.
- 2744 Clearspan Redundancy Guide.