Asterisk 10
Digium Partner Certification

Interoperability Report
ESCENE SayHi IP Phone

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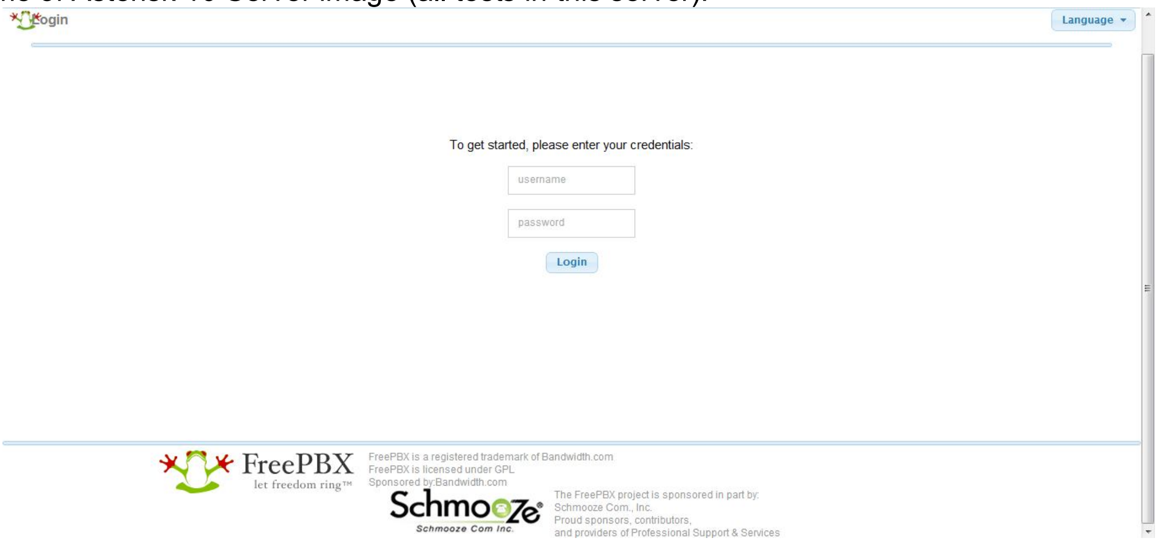
Directory

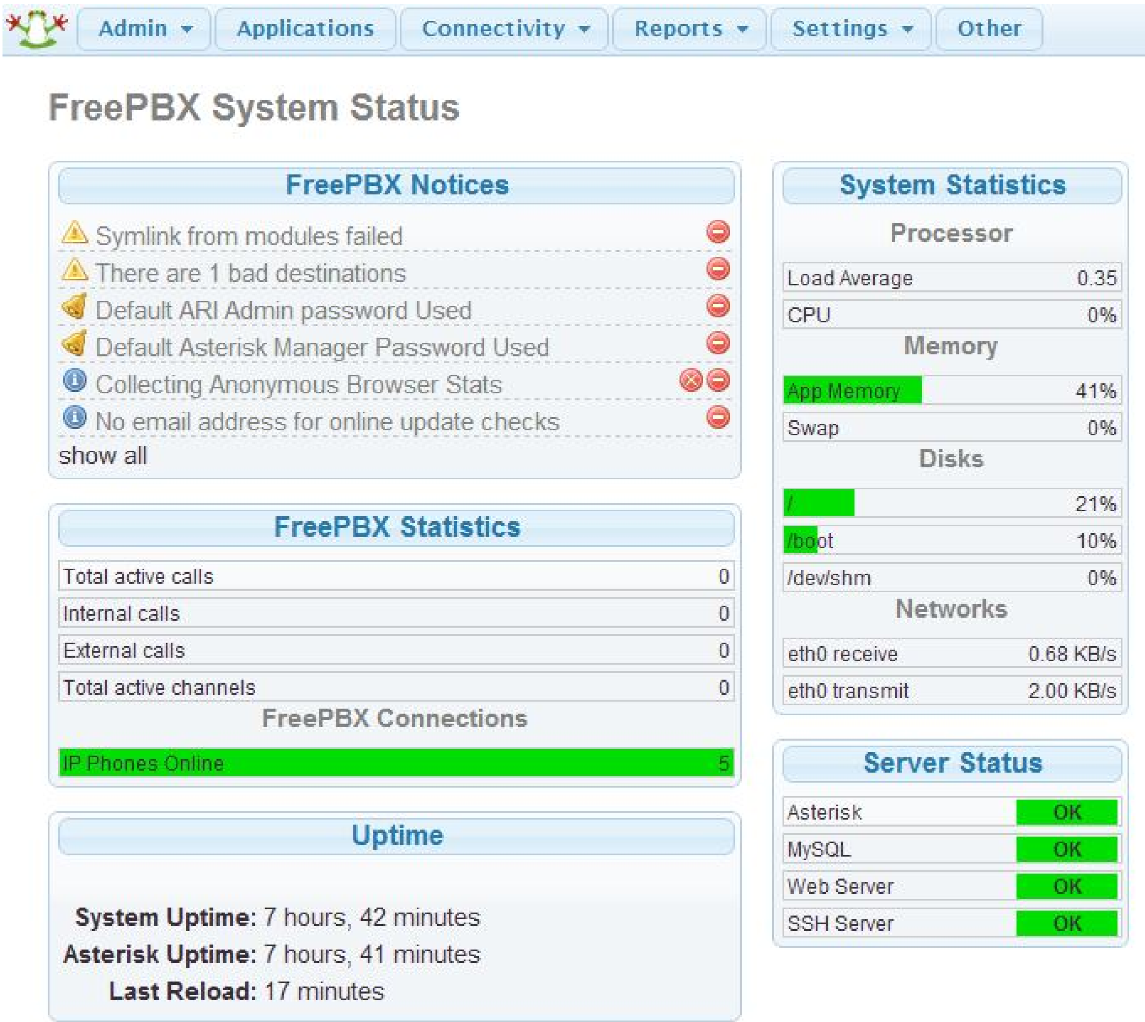
Subject :	**Interoperability Test Plan and Results of ESCENE Communication Technology IP Phones (CC800\ES220N\ES290N\292N\ES320NV3\ES330NV3\ES410\ES620) with** Asterisk 10	Date :	**May,06,2013**
revision history:		From:	Vinci Sam

EXECUTIVE SUMMARY

This document serves as the test plan and results for interoperability testing of ESCENE Communication Technology IP Phone CC800\ES220N\ES290N\292N\ES320NV3\ES330NV3\ES410\ES620 with Asterisk 10.

Some of Asterisk 10 Server image (all tests in this server):





1 INTRODUCTION

The focus of this testing is to verify that ESCENE IP Phones CC800\ES220N\ES290N\292N\ES320NV3\ES330NV3\ES410\ES620 can register to Session Manager and place calls to other endpoints via session manager.
Areas of testing include:

- Registration to Session Manager and other basic function.
- Audio Calls
- Long duration calls
- Codec negotiation
- Failure/Recovery

2 TEST COMPONENTS

Provider	Model	Software Version
Digium	Asterisk	10
ESCENE	SayHi IP Phone	Phone Model: CC800\ES220N\ES290N\292N\ES320NV3\ES330NV3\ES410\ES620 Software Version: v1.0.8.5-3870 Kernel Version: v2.6.4

3 TEST RESULTS

Complete the test report by marking an "X" in the appropriate column: **Pass**, **Fail**, **NA** (Not Applicable), **NS** (Not Supported), **NT** (Supported but Not Tested).

Test Case Title	Pass	Fail	NA	NS	NT
Basic Call Functionality	pass				
3-way Calling	pass				
Blind Call Transfer	pass				
Bulk Provisioning Using DHCP Option 66	pass				
Busy Lamp Field	pass				
Busy Lamp Field Pickup	pass				
Call Waiting	pass				
Call Name Display	pass				
Call Number Display	pass				
Codec G.711	pass				
Codec G.729	pass				
Codec G.722	pass				
Endpoint Voicemail - Basic Functionality	pass				
Call History	pass				
Do Not Disturb	pass				
Long Duration Calls	pass				

4 TEST CASES

4.1 Basic Call Functionality

Calls to/from the device can be setup and torn-down correctly. Run these tests with the **g711** codec if available on the device. These tests cover only 2-way calls with no services.

Test Case : Basic Endpoint Functionality - Test Case A	
Summary	Outgoing calls.
Setup	For SIP endpoints **authentication must be enabled if available**. For other devices, or for SIP devices that don't support authentication, mark the authenticated test as Approved Exception or Not Applicable. If the DUT is a trunk, application server or other device that does not itself have endpoints, then A and B must be known-good endpoints and the network must be set up so that calls between A and B are routed via the DUT.
Description	• Action: Dial B from A Confirm: B plays ring cadence Confirm: A plays ringback Action: Answer B Confirm: Two-way media is established Action: Hang up both phones Confirm: Pick up A and B to check both have dial tone
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

Test Case : Basic Endpoint Functionality - Test Case B	
Summary	Outgoing call canceled.
Setup	For SIP endpoints **authentication must be enabled if available**. For other devices, or for SIP devices that don't support authentication,

	mark the authenticated test as Approved Exception or Not Applicable. If the DUT is a trunk, application server or other device that does not itself have endpoints, then A and B must be known-good endpoints and the network must be set up so that calls between A and B are routed via the DUT.
Description	Action: Dial B from A Confirm: B rings Confirm: A plays ringback Action: Hang up A Confirm: B stops ringing Confirm: Pick up A and B, both have dial tone
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

Test Case : Basic Endpoint Functionality - Test Case C	
Summary	Incoming Call.
Setup	For SIP endpoints **authentication must be enabled if available**. For other devices, or for SIP devices that don't support authentication, mark the authenticated test as Approved Exception or Not Applicable. If the DUT is a trunk, application server or other device that does not itself have endpoints, then A and B must be known-good endpoints and the network must be set up so that calls between A and B are routed via the DUT.
Description	Action: Dial A from B Confirm: A rings Confirm: B plays ringback Action: Answer A

	Confirm: two-way media is established Action: Hang up both phones Confirm: Pick up A and B, both have dial tone.
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

Test Case : Basic Endpoint Functionality - Test Case D	
Summary	Incoming call cancelled
Setup	For SIP endpoints **authentication must be enabled if available**. For other devices, or for SIP devices that don't support authentication, mark the authenticated test as Approved Exception or Not Applicable. If the DUT is a trunk, application server or other device that does not itself have endpoints, then A and B must be known-good endpoints and the network must be set up so that calls between A and B are routed via the DUT.
Description	Action: Dial A from B Confirm: A rings Confirm: B plays ringback Action: Hang up B Confirm: A stops ringing Confirm: Pick up A and B, both have dial tone
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870

Author	Sam

Test Case : Basic Endpoint Functionality - Test Case E	
Summary	Long Duration Call
Setup	For SIP endpoints **authentication must be enabled if available**. For other devices, or for SIP devices that don't support authentication, mark the authenticated test as Approved Exception or Not Applicable. If the DUT is a trunk, application server or other device that does not itself have endpoints, then A and B must be known-good endpoints and the network must be set up so that calls between A and B are routed via the DUT.
Description	Action: Dial B from A Confirm: B rings Confirm: A plays ringback Action: Answer B Confirm: Two-way media is established Action: Maintain call for 5 minutes Confirm: Two-way media is still functioning Action: Hang up both phones Confirm: Pick up A and B, both have dial tone.
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

Test Case : Basic Endpoint Functionality - Test Case F	
Summary	User Signaling - Busy tone

Setup	For SIP endpoints **authentication must be enabled if available**. For other devices, or for SIP devices that don't support authentication, mark the authenticated test as Approved Exception or Not Applicable. If the DUT is a trunk, application server or other device that does not itself have endpoints, then A and B must be known-good endpoints and the network must be set up so that calls between A and B are routed via the DUT. A does not have call waiting or voicemail configured.
Description	Action: Take A off hook Action: Call A from B Confirm: A maintains dial tone Confirm: B plays busy tone. Action: Hang up both phones Confirm: Pick up A and B, both have dialtone.
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

Test Case : Basic Endpoint Functionality - Test Case G	
Summary	User Signaling - Dial tone timeout
Setup	For SIP endpoints **authentication must be enabled if available**. For other devices, or for SIP devices that don't support authentication, mark the authenticated test as Approved Exception or Not Applicable. If the DUT is a trunk, application server or other device that does not itself have endpoints, then A and B must be known-good endpoints and the network must be set up so that calls between A and B are routed via the DUT.
Description	Action: Take A off hook Confirm: A has dial tone Action: Leave A off hook for a prolonged period of time

	Confirm: Either: a) Dial tone stops and device can receive calls on that line again b) Device plays howler tone to replace handset
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam
Test Case : Basic Endpoint Functionality - Test Case G	
Summary	Do Not Disturb
Setup	For SIP endpoints **authentication must be enabled if available**. For other devices, or for SIP devices that don't support authentication, mark the authenticated test as Approved Exception or Not Applicable. If the DUT is a trunk, application server or other device that does not itself have endpoints, then A and B must be known-good endpoints and the network must be set up so that calls between A and B are routed via the DUT. DND may be done on the switch, or on the phone.
Description	Action: Set A to Do Not Disturb Action: Call A from B Confirm: Call is not established (for example: busy tone, voicemail, forwarded) Action: Hang up both phones Confirm: Pick up A and B, both have dial tone.
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

4.2 3-Way Calling

The device supports 3-way calling. The media for the calls may be mixed together by the device or by the Asterisk 10.

Test Case : 3-way calling - Test Case A	
Summary	3-way call
Setup	A is on the DUT B and C are known goods. A must have 3-way calling configured. For SIP lines, this requires multiple call appearances.
Description	Action: Setup call, B calling A. Action: Put B on hold and call C from A Confirm: 2-way media between C and A Confirm: 0-way media between B and A Action: Conference B in to form a 3 way call Confirm: 2-way media between all parties Action: repeat test, starting with A calling B instead.
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

4.3 Blind Call Transfer

The device supports blind call transfer, where a caller is transferred to a third party without the transferrer first connecting a second call with the third party

Test Case : Blind Call Transfer - Test Case A	
Summary	Successful Blind Call Transfer
Setup	A is on DUT B and C are on known goods. A and B are configured for Music on Hold. Note IP PBX DUTs: This feature should work if SIP REFER is used or not. It is suggested to test both modes.

	Call transfer must be enabled on line A.
Description	Action: Create a call from B to A Action: Blind transfer B to C Confirm: B hears Music on Hold during any on-hold period. Confirm: there is no media path A to C Confirm: A drops out of call Confirm: two way media between B and C Action: Hang up all phones
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

Test Case : Blind Call Transfer - Test Case B	
Summary	Failed Blind Call Transfer
Setup	A is on DUT B and C are on known goods. A and B are configured for Music on Hold. Note IP PBX DUTs: This feature should work if SIP REFER is used or not. It is suggested to test both modes. D is another Known Good. C must have voicemail and call waiting disabled.
Description	Action: Create a call from B to A Action: Create a call from C to D Action: From A attempt to blind transfer B to C Confirm: B hears Music on Hold during any on-hold period. Confirm: C and D remain in call with 2-way media Confirm: A and B reconnect with two-way media (note - when using POTS lines, you may need to flashhook once or twice more to get back to the original 2-way call, and you may get a 3-way call with the

	engaged tone during this process). Action: Hang up C and D Action: From A attempt to blind transfer B to C Confirm: B connected to C, with A disconnected Action: Hang up all phones
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

4.4 Bulk Provisioning Using DHCP Option 66

Some devices can receive an initial configuration from a TFTP server without any prior configuration. DHCP option 66 specifies the TFTP server name. This can be useful when deploying large quantities of phones.

Test Case : Bulk Provisioning Using DHCP Option 66 - Test Case A	
Summary	Bulk Provisioning Using DHCP Option 66 - Test Case A
Setup	You need a TFTP server to provide the initial configuration file and a DHCP server specifying your TFTP server address in option 66. TFTPD32 (http://tftpd32.jounin.net/) can provide both of these functions. DUT is at factory default settings. TFTP server contains a correctly named configuration file which points DUT at a Asterisk 10 SIP Provisioning Server.
Description	Action: Allow DUT to get a DHCP lease from your server providing option 66. Confirm: DUT then downloads initial configuration from TFTP server. Confirm: DUT is configured to download configurations from the provisioning server.
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-

	3870
Author	Sam

Test Case : Bulk Provisioning Using DHCP Option 66 - Test Case B	
Summary	Ignoring Option 66 After Initial Provisioning
Setup	You need a TFTP server to provide the initial configuration file and a DHCP server specifying your TFTP server address in option 66. TFTPD32 (http://tftpd32.jounin.net/) can provide both of these functions. DUT has already loaded initial configuration from TFTP server.
Description	Action: Reboot DUT, allowing it to get a DHCP lease from the server with option 66. Confirm: DUT does not download configuration file from TFTP server.
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

4.5 Busy Lamp Field

The device is capable of subscribing for more than one Busy Lamp Field (BLF, also sometimes referred to as "Line State Monitoring" or LSM). If the device can only subscribe to one busy lamp field, this feature should be marked "Verified as unsupported"

Test Case : Busy Lamp Field - Test Case A	
Summary	Subscribing to monitor a line
Setup	A is a business group line on the DUT and has line state monitoring enabled. B is on a Known Good and belongs to the same business group as A C is on a Known Good and does not belong to the same Business Group as A and B.
Description	Action: Configure A to monitor B Confirm: Check the call flows to confirm A's request to the Asterisk to monitor B is accepted correctly (200OK to SUBSCRIBE followed by a

	NOTIFY to give current state).
Test Result	Pass, but the CC800\ES220\290\292 mode didnt have programmable key (Not support)
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

Test Case : Busy Lamp Field - Test Case B	
Summary	Subscribing to monitor an unavailable line
Setup	A is a business group line on the DUT and has line state monitoring enabled. B is on a Known Good and belongs to the same business group as A C is on a Known Good and does not belong to the same Business Group as A and B.
Description	Action: Configure A to monitor C Confirm: Check the call flows to confirm A's request to the Asterisk to monitor C is rejected correctly (403 forbidden).
Test Result	Pass, but the CC800\ES220\290\292 mode didnt have programmable key (Not support)
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

Test Case : Busy Lamp Field - Test Case C	
Summary	Monitoring lines in ringing and in-call states
Setup	A is a business group line on the DUT and has line state monitoring enabled. B is on a Known Good and belongs to the same business group as A C is on a Known Good and does not belong to the same Business Group as A and B.
Description	Action: Configure A to monitor B

	Action: Ring B from C Confirm: A shows that B is ringing Action: Answer B Confirm: A shows that B is in a call Action: Hang up B, C. Confirm: A shows that B is idle.
Test Result	Pass, but the CC800\ES220\290\292 mode didnt have programmable key (Not support)
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

4.6 Busy Lamp Field Pickup

When monitoring other lines using the Busy Lamp Field features, this device can answer calls on the monitored lines. Note that this does not include "barge-in", which is when the monitoring device can take an already active call.

Test Case : Busy Lamp Field Pickup - Test Case A	
Summary	Pickup of a monitored line
Setup	A is a business group line on the DUT and has line state monitoring enabled. B is on a Known Good and belongs to the same business group as A.
Description	Action: Call B from C Confirm: A indicates that B is in a 'ringing' state Action: Answer C from A Confirm: Two-way media between C and A Confirm: B stops ringing Action: Hang up all phones

	Confirm: Pick up A, B, C, get dialtone on all.
Test Result	Pass, but the CC800\ES220\290\292 mode didnt have programmable key (Not support)
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

4.7 Call Waiting

The device supports call waiting, where a caller is notified during an active call of another inbound call.

Test Case : Call Waiting - Test Case A	
Summary	Call Waiting
Setup	A is on DUT B and C are on known goods. A has call waiting provisioned. For most SIP lines, this means setting multiple call appearances on the switch - at least 2 call appearances are required for call waiting.
Description	Action: Create a call from B to A Action: Call A from C Confirm: A indicates call waiting, by audible or visual notification (or both) Action: Put B on hold to take call C Confirm: 0-way media A to B Confirm: 2-way media A to C Action: Put C on hold to resume call with B Confirm: 0-way media A to C Confirm: 2-way media A to B

Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

Test Case : Call Waiting - Test Case B	
Summary	Call Waiting ignored
Setup	A is on DUT B and C are on known goods. A has call waiting provisioned
Description	Action: Create a call from B to A Action: Call A from C Confirm: A indicates call waiting, by audible or visual notification (or both) Action: Ignore call waiting, maintain call A-B Confirm: 2-way media remains A to B Confirm: 0-way media A to C Confirm: call waiting signal repeats for A Action: hang up C Confirm: call waiting signal ceases
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

4.8 Calling Name Display

The device can display the calling party name while the call is in ringing state.

Test Case : Calling Name Display - Test Case A	
Summary	isplay Calling Name on DUT
Setup	A is on DUT. B is a known good and does not have caller ID blocking enabled. B has a calling name configured either locally (for on-switch calls) or using a lookup service for off-switch callers.
Description	Action: B calls A Confirm: B's caller name is displayed on A's endpoint device
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

4.9 Calling Number Display

The device can display the calling party number while the call is in ringing state.

Test Case : Calling Number Display - Test Case A	
Summary	Calling number displayed
Setup	A is on DUT. B is a known good and does not have caller ID blocking enabled.
Description	Action: B calls A. Confirm: while A is ringing, user can see B's number.
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

4.10 Codec G.711

The device supports receiving and sending media using the G.711 Codec (either mu-law or A-law - agree this with your Asterisk contact).

Test Case : G.711 Codec - Test Case A	
Summary	G.711 codec negotiation led by DUT
Setup	Configure the DUT to support G.711 mu-law or A-law by checking the fixbits on the Remote Media Gateway Model on the Asterisk A is on DUT, and uses indirect media B is a Known Good
Description	Action: Call B from A Confirm: Check flows to confirm that G.711 has been negotiated correctly Confirm: two-way voice path Confirm: RTP media is sent in both directions using G.711 only.
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

Test Case : G.711 Codec - Test Case B	
Summary	G.711 codec negotiation led by Asterisk
Setup	Configure the DUT to support G.711 mu-law or A-law by checking the fixbits on the Remote Media Gateway Model on the Asterisk A is on DUT, and uses indirect media B is a Known Good
Description	Action: Call A from B Confirm: Check flows to confirm that G.711 has been negotiated correctly Confirm: two-way voice path Confirm: RTP media is sent in both directions using G.711 only.

Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

4.11 Codec G.729

The device supports receiving and sending media using the G.729 Codec.

Test Case : G.729 Codec - Test Case A	
Summary	G.729 codec negotiation led by DUT
Setup	Configure the DUT to support and prefer G.729 by checking the fixbits on the Remote Media Gateway Model on the Asterisk A is on DUT. B is a Known Good A is configured to use indirect media. If A is a SIP phone, the codec preference on the device must be set to prefer G.729, instead of changing the Remote Media Gateway Model fixbits.
Description	Action: Call B from A Confirm: Check flows to confirm that G.729 has been negotiated correctly Confirm: two-way voice path Confirm: RTP media sent over G.729 only in both directions
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

4.12 Codec G.722

The device supports sending and receiving media using G.722

Test Case : G.722 Codec - Test Case A	
Summary	G.722 negotiation led by known good

Setup	Configure the DUT to support and prefer G.722 Ensure that the Asterisk Remote Media Gateway model for this device has been set up to allow advanced codecs. You may need to contact your ITG support person for this. A is on DUT B is a known good
Description	Action: Call A from B Confirm: Check call setup flows to determine that G.722 is correctly negotiated Confirm: Two way media Confirm: RTP media is sent in both directions using G.722
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

Test Case : G.722 Codec - Test Case B	
Summary	G.722 codec negotiation led by DUT
Setup	Configure the DUT to support and prefer G.722 Ensure that the Asterisk Remote Media Gateway model for this device has been set up to allow advanced codecs. You may need to contact your ITG support person for this. A is on DUT B is a known good
Description	Action: Call B from A Confirm: Check call setup flows to determine that G.722 is correctly negotiated Confirm: Two way media Confirm: RTP media is sent in both directions using G.722
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870

Author	Sam

4.13 Endpoint Voicemail - Basic Functionality

In a network where a voicemail server is available, the device is able to leave, pickup and signal voicemail messages correctly.

Test Case : Endpoint Voicemail - Basic Functionality - Test Case A	
Summary	Leaving and Retrieving Voicemail
Setup	A is on DUT B is on Known Good A voicemail server is configured on the Asterisk and A is provisioned with a mailbox.
Description	Action: Dial A from B. Do not answer A Confirm: After a defined period of time A stops ringing and the call is forwarded to voicemail. Action: Leave a voicemail for A from B Action: Hang up B. Action: Take A off hook Confirm: A signals that a message is waiting (audible and/or visual) Action: Dial the voicemail retrieval number (e.g. *97) from A Action: Listen to and delete the message Action: Hang up A and then go back off-hook Confirm: A no longer signals message waiting Action: Hang up all phones
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

4.14 Call History

This page allows users to retrieve, store, and view call history data from the phone. Display information for each call includes the date and time when the call is made/received, and the phone number or contact name of the call.

Test Case : Call History	
Summary	This test verifies the functionality of the Call History feature.
Setup	1. Using the phone LCD menu navigation, clear the Call History records in the UUT. Note that most phones have history for: Placed Calls, Received Calls, and Missed Calls. Some phones with limited feature sets may only have history for: Placed Calls and Received Calls. 2. Place a call from UUT to Phone A, then answer the call and hangup. 3. Place a call to UUT from Phone A, then answer the call and hangup. 4. Place a call to the UUT, then let it go to VoiceMail. 5. Check the Call History in the UUT.
Description	- All Call History records will be cleared from the phone. - The Call History in the UUT will show: - One call placed by the UUT to Phone A - One call received by the UUT from Phone A - One missed call from Phone A
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

4.15 Do Not Disturb

Test Case : Do Not Disturb	
Summary	This test verifies the functionality of the Do Not Disturb feature.
Setup	1. Place a call from Phone A to the UUT. 2. End the call. 3. Select Do Not Disturb on the UUT. 4. Place a call from Phone A to the UUT.

	5. Disable Do Not Disturb on the UUT. 6. Place a call from Phone A to the UUT.
Description	- UUT rings, then a two-way voice path will be established when the UUT is answered. - UUT will not ring and the call will go to VoiceMail. - A two-way voice path will be established from the UUT to Phone A.
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

4.16 Long Duration Calls

Test Case : Long duration calls	
Summary	This test verifies the functionality of the Long duration calls feature.
Setup	A is DUT and B is good. Registered to server, then A and B must be known-good endpoints and the network must be set up so that calls between A and B are routed via the DUT.
Description	Action: Setup call, B calling A. Confirm: Two-way media is established Confirm: Long duration calls 24^{hr}. Action: Hang up both phones
Test Result	Pass
Test Notes	Test performed on Build ESCENE IP Phone CC800\US102\CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620-v1.0.8.5-3870
Author	Sam

5 CONCLUSION

The compliance testing of ESCENE Communication Technology IP Phones (CC800\US102\ES220\ES290\292\ES320V3\ES330V3\ES410\ES620) with Digium has been completed successfully. All test cases passed except for the observations mentioned in station**3.**