XO® SIP Service

Customer Configuration Guide

for CISCO 2821 with CME and CUE

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Table of Contents

1. Overview

About This Document

This document contains the test results of the SIP Trunking evaluation performed on the Cisco 2821 with Call Manager Express (CME) and Cisco Unity Express (CUE). **Note:** The software and the configuration for the Cisco 2821 and 2811 is the same. The only area where the units are different is in the hardware performance area.

- The Cisco 2821 was connected to the Juniper Networks ERX 1400 via 3 T1s configured using Multi-Link Point-to-Point Protocol (MLPPP).
- Cisco 7960G IP phones running Skinny Client Control Protocol (SCCP) are connected to a Catalyst 3560 Power over Ethernet (PoE) switch, which is up linked to the Cisco 2821 via a Gigabit Ethernet port 0/0.
- Data is connected to the Cisco 2821 via Gigabit Ethernet port 0/1.
- It is expected that the customer LAN will connect directly to the Cisco 2821 and not the Cisco 3560 PoE switch, which is reserved only for Cisco 7960G IP phones.

Executive Summary

This report provides the test results found to date for the SIP Trunking evaluation performed on the Cisco 2821 with CME and CUE.

- The Cisco IOS and CME software changed from version advipservices 124-11.XJ3 to advipservices 124-15.T for the IOS, 124-11XJ3 to 124-15.T for the CME, and the CUE software remained the same as listed in Section 2. Equipment Requirements for Testing, Item 1.d.
- The Sonus NBS software changed from 6.3.8 to 6.4.6, and the BroadSoft software remained at version 13.0.

The Broadband Data Engineering Lab performed a full regression test evaluation using service packages 1 and 2, verifying all test cases through the Sonus NBS. The following is a summary of the service affecting issues found to date:

- Cisco ticket number 606712793 was opened to address a service affecting issue where the Cisco 2821 reboots after a WAN link outage. Although Cisco has a software patch for this problem, the patch cannot be downloaded via Cisco's website, and it has not been incorporated into an official supported release.
- Cisco ticket number 606732447 was opened to address a service affecting issue where **voice prioritization o**ver data does not work properly using test configuration 1 or 2. Voice prioritization test results are shown in Section 4. Test Results, test case items 26 and 27. Additional troubleshooting time is required to isolate the voice prioritization issue.

2. Equipment Requirements for Testing

The following table lists the software and hardware required for testing.

Item	Software / Hardware	Version / Release
1	Cisco 2821 with CME and CUE	
	a. Cisco 2821 Software	Version: c2800nm-advipservicesk9-mz. 124-15.T.bin
	b. CME Software	Version: cme-124-15.T
	c. Quick Configuration Tool (QCT) Software	2.0.2
	d. CUE Software running:	

Installed Packages	Version / Release
Installer	2.3.2
Thirdparty	2.3.2
Bootloader (Primary)	2.1.2
Infrastructure	2.3.3
Global	2.3.2
GPL Infrastructure	2.3.0
Voice Mail	2.3.3
Bootloader (Secondary)	2.1.2
Core	2.3.0
Auto Attendant	2.3.0
Installed Languages	**Version / Release**
English language pack	2.3.0

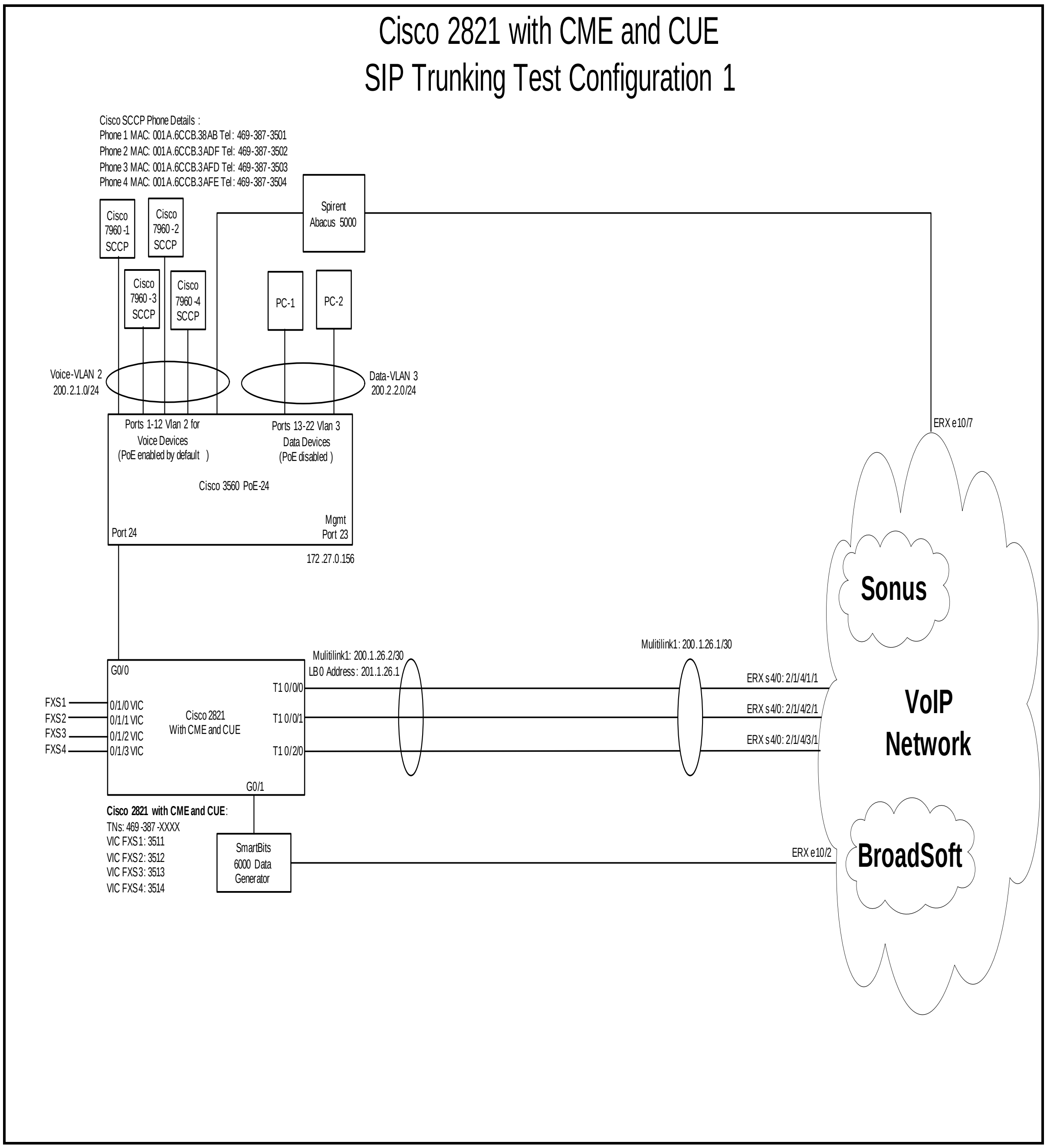
Item	Software / Hardware	Version / Release
2	Cisco 3560 PoE	c3560-ipbase-mz.122-25.SEE2.bin
3	Cisco 7960 G Phone	Software Version: P00308000400 equivalent to 8.0(4.0)
4	Cisco 2811 Used as a Router	Software Version: ipbase9-mz.124-4.T7
5	Juniper Networks ERX 1400	System Release: erx_6-1-3p0-11.rel
6	BroadSoft	System Release: 13.0
7	Sonus NBS	Release: V06.04.06R000
8	Sonus PSX and GSX	Release: 5.1
9	4 Analog Phones with Caller ID Displays	
10	All PBXs Approved for Use in the Flex Options II Service Offering	

3. Test Configurations

Configuration 1

Cisco 2821 Acting as an IP PBX and a Router

The following diagram is the configuration used during lab testing where the Cisco 2821 acts as an IP PBX and a router. Due to problems encountered during voice prioritization testing using Configuration 1, the diagram illustrated in Configuration 2 was used. Detailed results can be found in Section 4. Test Results.

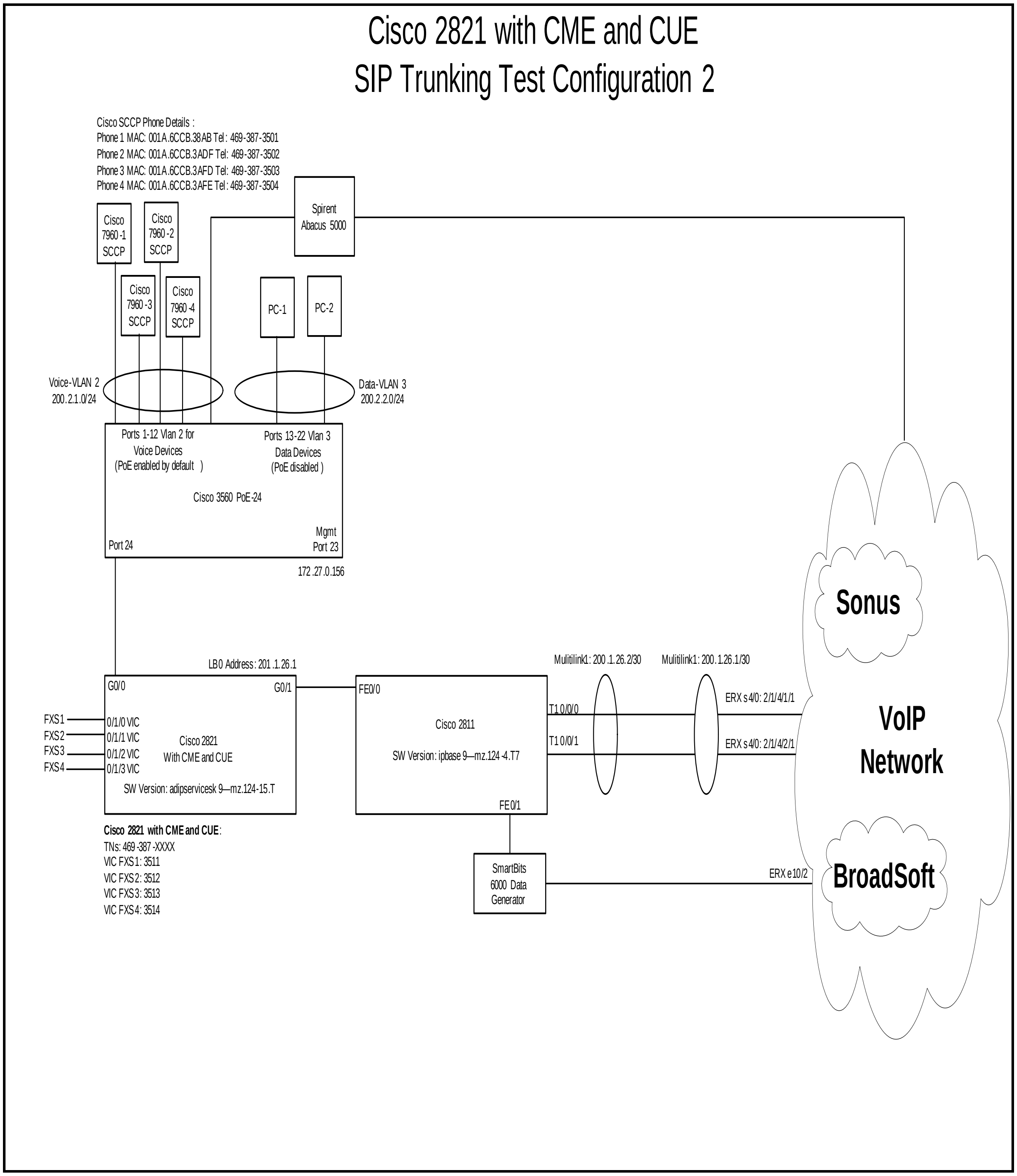


3. Test Configurations *(continued)*

Configuration 2

Cisco 2821 Acting Only as an IP PBX, and the Cisco 2811 Acting Only as a Router

The following diagram is the configuration used only to isolate voice prioritization issues encountered using Configuration 1. In Configuration 2, the Cisco 2821 is used only as an IP PBX, and the data is processed by the Cisco 2811, used only as a router. Detailed results can be found in Section 4. Test Results.



4. Test Results

Results Tables 1-30

The following tables present the SIP Trunking test results for Cisco 2821 running CME and CUE.

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
1	**1. Dynamic Registration**	This test case group was verified using only service package 1 because the registration process is service package independent.		
1.1	Dynamic Registration with Authentication	Dynamic registration is successful only if the username and password on BroadSoft and the PBX are the same. Group level username and password is supported by the PBX. The Registration Expire Timer range of the PBX is 60-65535 seconds, and the default is 3600 seconds.	Pass	Pass
1.2	Unregister User	Outbound calls from unregistered users are denied by the BroadSoft AS.	Pass	Pass
1.3	Min Registration Timer	The PBX minimum registration timer range is 60 to 65535 seconds. With the PBX expire timer configured for a minimum allowed value of 60 seconds, the expire timer returned from NBS 200 OK is 3597 seconds, and the AS 200 OK is 3599 seconds. The Sonus NBS responds to subsequent registration messages without forwarding them to AS until the timer between the Sonus NBS and BroadSoft AS has expired.	Pass	Pass
1.4	Max Registration Timer	With the PBX expire timer configured for a maximum allowed value of 65535 seconds, the expire timer returned from the Sonus NBS 200 OK is 3600 seconds, and the AS 200 OK is 65534.	Pass	Pass
1.5	Registration Refresh	Removing the registrar configuration line from the PBX causes the PBX to send out a registration message for each registered user with the expire timer equal to 0, which clears out all registrations on the AS.	Pass	Pass

4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
2	**2. Basic Calls Verification**	**Extension refers to IP-based 4-digit extension.**		
2.1	Extension-to-Extension	Verified using 4-digit dialing.	Pass	Pass
2.2	From PSTN		Pass	Pass
2.3	10-digits to PSTN		Pass	Pass
2.4	1+10-digits to PSTN		Pass	Pass
2.5	Inbound Toll Free		NA	NA
2.6	Outbound Toll Free	800-433-7300	Pass	Pass
2.7	Operator 0-		Pass	Pass
2.8	Operator 0+10-digits	The call was sent to the PSTN destination directly instead of sending it as an operator call. This appears to be a Sonus 5.1 configuration issue.	**Fail**	**Fail**
2.9	International Operator 01+	A fast busy is heard on the originating side of the call. This may also be a Sonus 5.1 configuration issue.	**Fail**	**Fail**
2.10	International 011+IDDD		NT	NT
2.11	411		Pass	Pass
2.12	911		NT	NT
2.13	Extension to Flex II MGCP phone		Pass	Pass
2.14	Flex II MGCP phone to Extension		Pass	Pass
2.15	PBX-to-PBX call (two groups in the same enterprise)	Verified Inter-Group calls work properly between two PBXs of the same enterprise using BroadSoft Voice-VPN.	Pass	Pass
2.16	PBX-to-PBX call (two groups of different PBX manufacturers)	Verified inter-enterprise calls using Avaya IP Office 500 to the CME 2811 in a different enterprise	Pass	Pass
2.17	Call to ACME Packet SD II SBC		Pass	Pass
2.18	Call from ACME Packet SD II SBC		Pass	Pass

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4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
3	**3. Caller ID (CLID) and Calling Name (CNAM) Presentation**			
3.1	Inbound CLID from PSTN	For this test '2806 Compliance' must be disabled on the Sonus NBS.	Pass	Pass
3.2	Out to PSTN	Only the CLID was displayed	Pass	Pass
3.4	Inbound Caller ID Blocking	Verify the PBX can mask the inbound CLID as requested by any PSTN caller where the caller dials *67xxxxxxxxxx.	Pass	Pass
3.5	Outbound Caller ID Blocking	This test case was verified by using the PBX feature, not the AS feature.	Pass	Pass
3.6	Inbound CNAM	Verified inbound CNAM from a BroadSoft and PSTN user.	Pass	Pass
3.7	Outbound CNAM	Verified outbound CNAM to a BroadSoft and PSTN user.	Pass	Pass

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4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
4	**4. Call Forward** Call Forward: (1st party = calling; 2nd party = 1st called party; 3rd party = the call forwarded to party)	**Verified bi-directional RTP path for a forwarded call. WAN bandwidth consumption for a forwarded call was not checked.**		
4.1	Call Forward Always to Extension from PSTN	Called from PSTN to an extension and always forwarded to another extension without 302 redirection. The call was forwarded within CME transparent to Sonus NBS.	Pass	Pass
4.2	Call Forward Always to PSTN from PSTN	Called from PSTN to an extension and always forwarded to another PSTN number. Bi-directional RTP path was verified. This test case now passes with Sonus 6.4.6 load where previously it failed due to the Sonus 0.0.0.0 bug.	Pass	Pass
4.3	Call Forward Busy to PSTN from PSTN	Called from PSTN to an extension and forwarded on busy to another PSTN number. Verify bi-directional RTP path.	Pass	Pass
4.4	Call Forward No answer to PSTN from PSTN	Called from PSTN to an extension and forwarded on no-answer to another PSTN number. Bi-directional RTP path was verified. This test case now passes with Sonus 6.4.6 load where previously it failed due to the Sonus 0.0.0.0 bug.	Pass	Pass
4.5	Call Forward No answer to PSTN From Extension	Called from one extension to another extension and forwarded on no-answer to a PSTN number. This test case now passes with Sonus 6.4.6 load where previously it failed due to the Sonus 0.0.0.0 bug.	Pass	Pass

4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
5	**5. Call Transfer** Call Transfer: 1st party = calling party 2nd party = 1st called party 3rd party = the call transferred-to party	**Verified bi-directional RTP path. WAN bandwidth consumption for a 'TRANSFERRED' call was not checked. Blind and consult transfer modes were tested. Leading 9 is used for PSTN transferred calls. No REFER messages were used.**		
5.1	Call Transfer to extension – blind	Called from PSTN to an extension and transferred to another extension (blind transfer). CME masked the internal transfer from network.	Pass	Pass
5.2	Call Transfer to extension – consult	Called from PSTN to an extension and transferred to another extension (consult transfer).	Pass	Pass
5.3	Call Transfer to PSTN – blind	The call does not establish after the transfer to the second PSTN number.	**Fail**	**Fail**
5.4	Call Transfer to PSTN – consult	Called from PSTN to an extension and transferred to another PSTN number using consult transfer. Bi-directional RTP path was verified. RTP is hair pinned through CME. The call stays G.729 all the way, and no transcoding on the CME is used. Note: Although the call works properly, the CME call ID is displayed when the call is transferred to the PSTN phone, instead of the original call ID.	**Fail**	**Fail**
5.5	Call Transfer from extension to PSTN – blind	Called from extension to an extension and transferred to another PSTN number, using blind transfer. Bi-directional RTP path was verified.	Pass	Pass
5.6	Call Transfer extension to PSTN – consult	Called from extension to an extension and transferred to another PSTN number using consult transfer. Bi-directional RTP path was verified. RTP is hair pinned through CME where the extension-to-extension call is G.711, and then it is transcoded internally within CME to G.729 to PSTN.	Pass	Pass

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4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
6	**6. Analog – FXS Port Verification**	**One page faxes were used for these tests, and the fax copies received were legible. G3 fax machines were used for these tests.**		
6.1	Inbound fax - G711	The CME sends a reINVITE with G.711, and the Sonus NBS responds with another T.38 reINVITE. The CME rejects it and sends an ACK to the Sonus NBS. The CME is configured as fax using G711 pass-through.	Pass	Pass
6.2	Outbound fax - G711	The Sonus NBS sends a reINVITE with T.38. The CME responds with G711, which the Sonus NBS accepts. The CME is configured as fax using G711 pass-through.	Pass	Pass
6.3	Inbound fax T.38	The CME send a reINVITE with T.38, which the Sonus NBS accepts. The CME is configured for as T.38 fax.	Pass	Pass
6.4	Outbound fax -T.38	The Sonus NBS sends a reINVITE with T.38, which the CME accepts. The CME is configured for as T.38 fax.	Pass	Pass
6.5	Analog Phone to IP-extension	4-digit dialing was used for this test.	Pass	Pass
6.6	IP-extension to Analog Phone	4-digit dialing was used for this test.	Pass	Pass
6.7	Analog Phone to PSTN		Pass	Pass
6.8	PSTN to Analog Phone		Pass	Pass
6.9	Inbound Modem (G.711)	Ticket number 606594289 has been opened with Cisco. This issue is not service-package related.	**Fail**	**Fail**
6.10	Outbound Modem (G.711)	Ticket number 606594289 has been opened with Cisco. This issue is not service-package related.	**Fail**	**Fail**

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4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
7	**7. Dual Tone Multi-Frequency (DTMF) Tests**	**DTMF capabilities of PBXs varies, thus not all tests are applicable.**		
7.1	Outbound RFC2833 to PSTN	Called American Airlines 800-433-7300 to verify DTMF tones.	Pass	Pass
7.2	Inbound RFC2833	Called the CME voice mail number to verify DTMF tones. This is the same as the CUE voice mail test.	Pass	Pass
7.3	Outbound in-band RTP DTMF to PSTN		NA	NA
7.4	Inbound in-band RTP DTMF	Access PBX voice mail portal from PSTN	NA	NA
7.5	Outbound SIP-INFO DTMF to PSTN	This feature will not be offered in the initial product launch. Test item is kept as a placeholder.	NA	NA
7.6	Inbound SIP-INFO DTMF	This feature will not be offered in the initial product launch. Test item is kept as a placeholder. Access PBX voice mail portal from PSTN.	NA	NA
8	**8. Three-way Conference Calls (PBX)**	**Two extensions and 1 PSTN party.** **Bi-directional RTP path was verified. WAN bandwidth consumption was not verified.**		
8.1	3rd party is extension	Called from PSTN to extension and conferenced in another extension	Pass	Pass
8.2	Third party is PSTN	Called from PSTN to extension 0291 and conferenced in a PSTN party 972-578-6572	Pass	Pass
8.3	Third party is extension of another package	Called from PSTN to extension of package 1 and conferenced in another extension on another PBX using package 2. Bi-directional RTP and voice quality were verified.	Pass	Pass
8.4	Call from an extension on CME1 2821 configured to use service package 2 to the PSTN and transfer the call to CME2 2811 configured to use service package 1.	Bi-directional RTP and voice quality were verified.	Pass	Pass

4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
9	**9. Acct/Auth Codes (AS)**			
9.1	Account Code on AS	BroadSoft does not verify that a valid account code is entered. In BroadSoft Release 13.0, the user is prompted to enter an account or authorization code only for calls outside the enterprise. The CME 2811 configured in a different enterprise was used for these tests.	Pass	Pass
9.2	Authorization Code on AS	BroadSoft verifies that a valid authorization code is entered. In BroadSoft Release 13.0, the user is prompted to enter an account or authorization code only for calls outside the enterprise. The CME 2811 configured in a different enterprise was used for these tests.	Pass	Pass
9.3	Invalid Auth Code on AS	Verified that after entering an invalid account code the call is established with bi-directional RTP when the correct authorization code is entered.	Pass	Pass
10	**10. Call Center (CC) AS**			
10.1	Verify that an unanswered CC call, it is forwarded to a final destination number.	Called from PSTN to CC.	Pass	Pass
10.2	Agent busy	While the first agent is busy with a call, verify that a call from PSTN to CC is forwarded to next agent correctly. Note: The command "call-forward busy 3519" must be removed from the CME configuration for the call to go to the next agent.	Pass	Pass
10.3	Call queuing	Verified that a call in queue is connected to a live agent once any agent becomes available.	Pass	Pass
10.4	Extension to CC	Called from extension to CC and verified that a call can be established	Pass	Pass
10.5	Multiple Call Distribution Policies: Circular, Regular, Simultaneous, Uniform, Weighted	Circular, Regular, Simultaneous, Uniform, and Weighted call distribution policies were verified.	Pass	Pass
10.6	Music on hold		Pass	Pass
10.7	Overflow		NT	NT
10.8	Statistics		NT	NT

4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
11	**11. Auto Attendant (AA) AS**			
11.1	Dial by extension			
11.2	Dial by Name	Note: The CLID last name and first name must be used for dial by name to work properly.	Pass	Pass
11.3	Call queuing: Verify that an AA number can handle more than one call simultaneously.		Pass	Pass
11.4	Enterprise-wide AA from PSTN: Create two groups in an enterprise. Assign AA to group 1 of the enterprise and verify a PSTN call to the AA can reach group 2.	Verified dial by extension and dial by name. Note: The CLID last name and first name must be used for dial by name to work properly.	Pass	Pass
11.5	Enterprise-wide AA from extension of Group 2: Create two groups in an enterprise. Assign AA to group 1 of the enterprise and verify the extension of group 2 can reach AA for other group 2 extension or Group 1 extension.	Verified dial by extension and dial by name. Note: The CLID last name and first name must be used for dial by name to work properly.	Pass	Pass
11.6	Customizable Menu Options		Pass	Pass
11.7	Holiday Schedule		Pass	Pass
11.8	Transfer to Operator		Pass	Pass
11.9	Record Greeting Remotely		NT	NT
11.10.	Record User Name		NT	NT

4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
12	**12. BroadSoft Voice VPN (AS)**	**This test requires two PBXs configured as two different groups in the same enterprise on the AS.**		
12.1	PBX1 to PBX2	Location code plus extension dialing between two PBXs of the same Enterprise	Pass	Pass
12.2	extension of PBX1 can leave voice mail in PBX 2	Verified that the extension of PBX1 can call an extension of PBX2. Verified the extension of PBX1 can leave voice mail in PBX2 when voice mail answers. Verified MWI is turned on the Ext of PBX2.	Pass	Pass
12.3	extension of PBX1 can retrieve voice mail in PBX2	Verified the extension of PBX1 can be used to retrieve voice mail message of an extension belonging to PBX2. Note: CUE reports unknown caller before the voice mail message is played.	Pass: **Warning**	Pass: **Warning**
13	**13. Voice Mail PBX**	The integrated voice mail features of the PBX were verified. Bi-directional RTP was verified, but WAN bandwidth consumption was not verified.		
13.1	Leave voice mail from an extension: Verify that DTMF works properly.		Pass	Pass
13.2	Leave voice mail from PSTN: Verify that DTMF works properly.		Pass	Pass
13.3	Retrieve voice mail from extension: Verify that DTMF works properly.		Pass	Pass
13.4	Retrieve voice mail from PSTN: Verify that DTMF works properly.		Pass	Pass
13.5	MWI on/off operations	For all test cases in this section verify that the MWI indicator is lit after leaving a message and that the indicator is turned off when messages are deleted.	Pass	Pass

4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
14	**14. Service Package Configuration Tests** service package 1 (SP1) and service package 2 (SP2)	**For these test cases basic functionality of the first two packages and interoperability between packages was verified.** **Codec and DTMF selections were in line with definition of the corresponding package.** **Bi-directional RTP and voice quality was also verified.**		
14.1	PSTN to extension SP1		Pass	NA
14.2	extension SP1 to PSTN		Pass	NA
14.3	extension SP1 to extension (PKG2)		Pass	NA
14.4	extension SP2 to extension SP1		NA	Pass
14.5	DTMF from extension SP1 to extension SP2		Pass	NA
14.6	DTMF from extension SP2 to extension SP1		NA	Pass
14.7	DTMF SP1 to PSTN		Pass	NA
14.8	Inbound DTMF SP1 from PSTN		Pass	NA
14.9	PSTN to extension SP2 codec 1 using G.729	Verify that G.729 is selected when the IP PBX is configured to prefer G.729 over G.711.	NA	Pass
14.10	PSTN to extension SP2 codec 2 using G.711	Verify that G.711 is selected when the IP PBX is configured to prefer G.711over G.729. Note: This test case fails due to Sonus ticket number 35603.	NA	**Fail**
14.11	extension SP2 to PSTN	Verify that with IP PBX configured to send G.729 first followed by G.711 that G.729 is selected.	NA	Pass
14.12	extension SP2 to PSTN	Verify that with the IP PBX configured to send G.711 first followed by G.729 that G.711 is selected. Note: This test case fails due to Sonus ticket number 35603.	NA	**Fail**
14.13	DTMF SP2 to PSTN		NA	Pass
14.14	PSTN to DTMF SP2		NA	Pass

4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
15	**15. BroadSoft AS Redundancy Support**	**Verify the level of impact on calls from and to the PBX during a primary AS failure. Shutdown the primary AS server after verifying all DNs have registered successfully. The REGISTRAR-MIN-EXPIRE timer on the Sonus NBS was configured for a smaller value to speed up these tests.**		
15.1	Registration: Verify that new registrations are forwarded to the secondary AS through the Sonus NBS and accepted successfully.	Note: The black listing feature is available on the Sonus NBS only for INVITE messages not REGISTER messages. The Sonus NBS was observed to send a REGISTER message to the primary AS and when no response was received the REGISTER message was sent to the secondary AS. This will occur for every REGISTER message until an INVITE is received which causes the Sonus NBS to black list the primary AS.	**Fail**	**Fail**
15.2	Active call: Note the impact on an active call when primary AS fails	There is no impact to an active call when RTP has already been established as the bi-directional RTP stream does not traverse BroadSoft. If the server goes down in the middle of call setup, the user may need to re-dial the number, or the equipment may send a re-INVITE.	Pass	Pass
15.3	Inbound call: Verify that inbound calls to the PBX are routed through to the secondary AS and work correctly.	The first INVITE that is received triggers the Sonus NBS to black list the primary AS. A small delay is perceived on the first call, and all other calls work normally.	Pass	Pass
15.4	Outbound call: Verify that outbound calls to the PBX are routed through to the secondary AS and work correctly.	The first INVITE that is received triggers the Sonus NBS to black list the primary AS. A small delay is perceived on the first call and all other calls work normally.	Pass	Pass
16	**16. NBS Redundancy Support**	**These test cases were verified using only service package 2 (SP2) because the NBS redundancy feature is independent of the service package used. The level of impact on calls from and to the PBX was verified during the primary PNS card failure.**		
16.1	Active call	Verified that the bi-directional RTP path is maintained during failover.	Pass	Pass
16.2	Inbound call	Verified that inbound calls to PBX are routed through the standby PNS and work correctly.	Pass	Pass
16.3	Outbound call	Verified that outbound calls to PBX are routed through standby PNS and work correctly.	Pass	Pass

4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
17	**17. Session Timer (re-INVITE)**	**Verify the PBX and NBS can negotiate Session Timer value and process RE-INVITEs correctly. Session Timer should be enabled towards PBXs and disabled towards AS. The Session Timer was reduced to a small value to speed up these test cases.** **Note that the tests were conducted using service package 2; however, these test cases are service package independent.**		
17.1	MIN-SE: Record the default MIN-SE value of the PBX (by checking the MIN-SE value in the INVITE message for outbound calls). Record if the PBX originates SESSION-EXPIRE request (by checking if SESSION-EXPIRE header is presence in the INVITE message for outbound calls). Configure the MIN-SE on PBX to be 3600 seconds and make inbound call. Verify that the NBS can handle the '422' response from the PBX correctly.	The default MIN-SE value is 1800 seconds. The expires timer value is 180 seconds. The Sonus NBS sent a MIN-SE of 90 seconds, and the CME sent a 422 session timer too small message with a MIN-SE value of 3600 seconds. No problems were experienced with any calls due to changing the MIN-SE value.	Pass	Pass
17.2	Refresher: Record if the Sonus NBS is the negotiated REFRESHER for both inbound and outbound calls.	The refresher value is User Agent Client (UAC) for inbound and outbound calls.	Pass	Pass
17.3	re-INVITE: Keep a call active long enough (greater than half of the SESSION-EXPIRE timer) to verify the correct operation of re-INVITE messages.	The MIN-SE parameter was configured for the minimum value of 90 seconds. On an inbound call the CME sends an INVITE, receives a 200 OK and then sends an ACK message. On an outbound call, the CME receives an UPDATE message and sends a 200 OK.	Pass	Pass
17.4	Session Tear Down: Keep a call active; disconnect the WAN link and wait for N=SESSION-EXPIRE seconds, verify if the call is cleared by PBX, NBS, and AS.	Verified that the call is torn down on the CME, the Sonus NBS, and BroadSoft.	Pass	Pass

4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
18	**18. BroadSoft Call Admission Control (CAC), using Call Capacity**	**Verify Call Capacity control on the AS works seamlessly with the Sonus NBS and the PBX.**		
18.1	Inbound and outbound: Verify that inbound and outbound calls to and from PBXs are restricted per the 'call capacity' parameters on AS.		Pass	Pass
18.2	Max Call Capacity: Verify that the maximum number of calls on the PBX (including inbound and outbound) are restricted per the call capacity parameters on AS.		Pass	Pass
19	**19. Call Hold (PBX)**			
19.1	Call Hold to extension, dial extension: Call from the PSTN to an extension and then put the call on hold. Verify that the same DID can make a call to another extension after putting the first call on hold. Drop the second call and resume the first call. Verify the first call can be re-connected.	The CME did not generate any message to signal the call on hold to the original caller, but it does generate a tone to the caller.	Pass	Pass
19.2	Call Hold to extension, dial PSTN: Call from the PSTN to an extension and then put the call on hold. Verify that the same DID can make a call to PSTN after putting the first call on hold. Drop the second call, and resume the first call. Verify the first call can be re-connected.	The CME does not generate any message to signal the call on hold to the original caller, but it does generate a tone to the caller.	Pass	Pass

▶

4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
20	**20. Conferencing more than 3 parties (AS)**	**Conferencing provided by BroadSoft**		
20.1	3 party conferencing: Call from the PSTN to an extension of SP1 who in turns makes a conference call to an extension of SP2 on another PBX. Verify that the all 3 can be joined in a conference call.	The Cisco 7960 phone only allows a dual-line configuration per extension. A conference call with a maximum of 3 parties was established.	Pass	Pass
20.2	4 party conferencing: Call from the PSTN to an extension of SP1 who in turns makes a conference call to an extension of SP2 on another PBX. Verify that the first extension is still able to make another call to the PSTN for the same conference call.	It should be noted that the Cisco 7960 phone only allows a dual-line configuration per extension. A conference call with more than 3 parties cannot be established.	**Fail**	**Fail**
21	**21. Hunt Group (HG) PBX**	**HG feature provided by PBX. The voice mail timeout for the hunt group members was configured for a value greater than the hunt group timeout. Only the sequential hunt group feature was verified.**		
21.1	Sequential: Verify the HG members are called in sequence.		Pass	Pass
21.2	Verify that at the end of the hunt sequence the call can be forwarded to a PSTN number or other BroadSoft number.		Pass	Pass

▶

4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
21.3	Verify that at the end of the hunt group (HG) sequence the call can be forwarded to voice mail.	Ticket number 606563785 has been opened with Cisco. Two solutions were provided which require CUE configuration changes that need to be tested. This issue is not service-package related. **Option 1** 1) Create a new ephone-dn with an unused DN XXXX. 2) Set it to Call Forward Always (CFA) to Voicemail Pilot. 3) Set this DN XXXX as FINAL for the hunt pilot configuration. Final is the number where the call should be forwarded in the event no one in the hunt group answers. 4) Add this XXXX DN as an E.164 number to one of the existing mailboxes where you would like to collect voicemails. Say XXXX is added as E.164 number to DN 2301's voicemail box. Now the unanswered calls from the Hunt Pilot will go to 2301's mailbox. **Option 2** 1) Create a new ephone-dn with unused DN XXXX. 2) Set it to CFA to voice mail Pilot number. 3) Set this DN as FINAL for hunt pilot configuration. Final is the number that the call should be forwarded to in the event no one in the hunt group answers. 4) Create a new General Delivery Mailbox for DN XXXX and make one or more users (Existing mailbox users) members of this group so that they can check the voicemails left in this Group mailbox.	**Fail**	**Fail**
21.4	Verify the HG skips a busy member: Busy the first HG member who has its own voice mail. Verify that incoming call skips the busied member and does not go to its voice mail. DND can be activated on one of the HG members for this test.	The test case was verified using the DND feature and by busying the phone with an inbound or an outbound call.	Pass	Pass
21.5	Simultaneous calls to the HG Pilot	While the first incoming call to the HG is ringing, make a second call to the same HG number, and verify that the second call will skip over the ringing member and will ring the next available member of the HG.	Pass	Pass

4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
22	**22. Auto Attendant (PBX)**	**Auto Attendant feature provided by the PBX**		
22.1	Dial by extension from PSTN		Pass	Pass
22.2	Dial by name from PSTN	Ticket number 606565259 was opened with Cisco to address this issue. Additional test time is required to verify the necessary CUE configuration changes that are required for this feature to work properly.	**Fail**	**Fail**
23	**23. BroadSoft Trunk Group Verification**	**BroadSoft Trunk Group Call Capacity Enforcement Verification with VGE=4. For this test extension 3501 was configured for only inbound with maximum calls=1, maximum inbound=1 and maximum outbound=blank. Extension 3502 was configured for only outbound with maximum calls=1, maximum inbound=blank and maximum outbound=1. Extensions 3503 and 3504 were configured for 2-way with maximum calls=2, maximum inbound=2 and maximum outbound=2.**		
23.1	Verify maximum inbound calls		Pass	Pass
23.2	Verify inbound number can only receive 1 inbound call		Pass	Pass
23.3	Verify that an inbound only number can not make an outbound call	On an inbound only trunk group BroadSoft does not allow the "Maximum Active Outgoing Calls Allowed" parameter to be set to a value of 0 and enforces a minimum value of 1 outbound call. This may be by design for emergencies. Ticket number 51790 was opened to address this issue.	**Warning**	**Warning**
23.4	Verify maximum outbound calls		Pass	Pass
23.5	Verify outbound number can only make 1 outbound call		Pass	Pass
23.6	Verify outbound number can not receive an inbound call	On an outbound only trunk group BroadSoft does not allow the "Maximum Active Incoming Calls Allowed" parameter to be set to a value of 0 and enforces a minimum value of 1 inbound call. This may be by design for emergencies. Ticket number 51790 was opened to address this issue.	**Warning**	**Warning**

▶

4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
23.7	Verify maximum 2-way inbound calls	BroadSoft Trunk Group Call Capacity Enforcement Verification with VGE=2. For this test extension 3501 (dual-line) and 3502 (single-line) were configured for 2-way with maximum calls=2, maximum inbound=2 and maximum outbound=2.	Pass	Pass
23.8	Verify maximum 2-way outbound calls	BroadSoft Trunk Group Call Capacity Enforcement Verification with VGE=3. For this test extension 3501 (dual-line) and 3502 (single-line) were configured for 2-way with maximum calls=2, maximum inbound=2 and maximum outbound=2.	Pass	Pass
23.9	With 1 inbound and 1 outbound call, verify that the Maximum Active Calls configured for 2 is enforced.	BroadSoft Trunk Group Call Capacity Enforcement Verification with VGE=3. For this test extension 3501 (dual-line) and 3502 (single-line) were configured for 2-way with maximum calls=2, maximum inbound=2 and maximum outbound=2.	Pass	Pass
24	**24. BroadSoft Trunk Group Call Flow Verification**	**BroadSoft Trunk Group Call Capacity Enforcement Call Flow Verification**		
24.1	Exceed "Maximum Active Calls Allowed" on an inbound only trunk	The originating caller receives a "486 Busy Here" if the "Maximum Active Calls Allowed" is exceeded with an inbound call. The originating caller receives a "403 Forbidden" if the "Maximum Active Calls Allowed" is exceeded with an outbound call.	Pass	Pass
24.2	Exceed "Maximum Active Incoming Calls Allowed" for an inbound only trunk	The originating caller receives a "486 Busy Here" if the "Maximum Active Incoming Calls Allowed" is exceeded with an inbound call.	Pass	Pass
24.3	Exceed "Maximum Active Outgoing Calls Allowed" for an inbound only trunk	The originating caller receives a "403 Forbidden" if the "Maximum Active Outgoing Calls Allowed" is exceeded with an outbound call.	Pass	Pass
24.4	Attempt to establish an outbound call on an inbound only trunk	On an inbound only trunk group BroadSoft does not allow the "Maximum Active Outgoing Calls Allowed" parameter to be set to a value of 0 and enforces a minimum value of 1 outbound call. This may be by design for emergencies. Ticket number 51790 was opened to address this issue. The call flow looks normal since the call is allowed.	**Warning**	**Warning**

4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
24.5	Exceed Maximum Active Calls Allowed on an outbound only trunk	The originating caller receives a "486 Busy Here" if the "Maximum Active Calls Allowed" is exceeded with an inbound call. The originating caller receives a "403 Forbidden" if the "Maximum Active Calls Allowed" is exceeded with an outbound call.	Pass	Pass
24.6	Exceed Maximum Active Outgoing Calls for an outbound only trunk	The originating caller receives a "403 Forbidden" if the "Maximum Active Outgoing Calls Allowed" is exceeded with an outbound call.	Pass	Pass
24.7	Exceed Maximum Active Incoming Calls for an outbound only trunk	The originating caller receives a "486 Busy Here" if the "Maximum Active Incoming Calls Allowed" is exceeded with an inbound call.	Pass	Pass
24.8	Attempt to establish an inbound call on an outbound only trunk	On an outbound only trunk group BroadSoft does not allow the "Maximum Active Incoming Calls Allowed" parameter to be set to a value of 0 and enforces a minimum value of 1 inbound call. This may be by design for emergencies. Ticket number 51790 was opened to address this issue. The call flow looks normal since the call is allowed.	**Warning**	**Warning**
24.9	Exceed Maximum Active Calls Allowed set for 2 on a 2-way trunk with 1 inbound call and 1 outbound call	The originating caller receives a "403 Forbidden" if the "Maximum Active Calls Allowed" is exceeded with an outbound call. The originating caller receives a "486 Busy Here" if the "Maximum Active Calls Allowed" is exceeded with an inbound call.	Pass	Pass
24.10.	Exceed Maximum Active Incoming Calls for a 2-way trunk	The originating caller receives a "486 Busy Here" if the "Maximum Active Incoming Calls Allowed" is exceeded with an inbound call.	Pass	Pass
24.11	Exceed Maximum Active Outgoing Calls for a 2-way trunk	The originating caller receives a "403 Forbidden" if the "Maximum Active Outgoing Calls Allowed" is exceeded with an outbound call.	Pass	Pass

4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
25	**25. Maximum Voice Grade Equivalent (VGE) Capacity**	**VGE Capacity Testing**		
25.1	Verify the maximum G.729 voice calls the Cisco 2821 can handle using service package 2	The maximum number of G.729 calls on 3 T1s is 72; however, the CME is limited to 48 actual handsets. The maximum VGEs tested is 48.	Pass	Pass
25.2	Verify the maximum G.711 voice calls the Cisco 2821 can handle using service package 1	The maximum number of G.729 calls on 3 T1s is 72; however, the CME is limited to 48 actual handsets. The maximum VGEs tested is 48.	Pass	Pass
26	**26. Voice Prioritization Using Test Configuration 1**	**Verify that voice is prioritized over data**		
26.1	Using test configuration 1 where the Cisco 2821 is running CME, CUE and router functions in the same device, verify that voice calls are prioritized over data using service package 1.	The maximum number of G.729 calls on 3 T1s is 72; however, the CME is limited to 48 actual handsets, thus the maximum VGEs tested is 48. Voice prioritization over data does not work properly. RTP packet loss is observed in the Cisco 2821 towards the ERX direction. Ticket number 606732447 has been opened with Cisco. Trouble-shooting is required to isolate this issue.	**Fail**	**Fail**
26.2	Using test configuration 1 where the Cisco 2821 is running CME, CUE and router functions in the same device, verify that voice calls are prioritized over data using service package 2.	The maximum number of G.729 calls on 3 T1s is 48. The CME is limited to 48 actual handsets, thus the maximum VGEs tested is 48. Test configuration 1 was used for this test case. Using test configuration 1 where the Cisco 2821 is running CME, CUE and router functions in the same device, voice prioritization over data does not work properly. RTP packet loss is observed in the Cisco 2821 towards the ERX direction. Ticket number 606732447 has been opened with Cisco. Trouble-shooting is required to isolate this issue.	**Fail**	**Fail**

4. Test Results *(continued)*

Results Tables 1-30 *(continued)*

Test Case Item	Description	Comments	Result Using Svc Pkg 1	Result Using Svc Pkg 2
27	**27. Voice Prioritization Using Test Configuration 2**	**Verify that voice is prioritized over data**		
27.1	Using test configuration 2 where the Cisco 2821 is running only CME and CUE functions in the same device and the Cisco 2811 is acting only as a router, verify that voice calls are prioritized over data using service package 1.	Using Test Configuration 2 where the Cisco 2821 only processes calls and the Cisco 2811 is used as a router there is still RTP packet loss observed in the Cisco 2821 towards the ERX direction. Ticket number 606732447 has been opened with Cisco. Trouble-shooting is required to isolate this issue.	**Fail**	**Fail**
27.2	Using test configuration 2 where the Cisco 2821 is running CME and CUE functions in the same device and the Cisco 2811 is acting only as a router, verify that voice calls are prioritized over data using service package 2.	Using Test Configuration 2 where the Cisco 2821 only processes calls and the Cisco 2811 is used as a router there is still RTP packet loss observed in the Cisco 2821 towards the ERX direction. Ticket number 606732447 has been opened with Cisco. Trouble-shooting is required to isolate this issue.	**Fail**	**Fail**
28	**28. Group Caller ID**	**Group Caller ID Function Verification**		
28.1	Verify Group Caller ID is passed to a different enterprise only	In BroadSoft Release 13.0, calls to different groups will only transmit the group calling ID if the call is going to a different enterprise.	Pass	Pass
28.2	Verify Group Name is passed to a different enterprise only	In BroadSoft Release 13.0, calls to different groups will only transmit the group-calling name if the call is going to a different enterprise.	Pass	Pass
29	**29. Incoming Calling Plans**			
29.1	Verify partially blocked outside calls	In BroadSoft Release 13.0, only calls from outside the enterprise are blocked.	Pass	Pass
29.2	Verify blocked outside calls	In BroadSoft Release 13.0, only calls from outside the enterprise are blocked.	Pass	Pass
30	**30. Call Detail Record (CDR)**			
30.1	Collect various CDRs for the IT department.	Extensive CDRs were gathered during BroadSoft 13.0 testing. CME specific CDRs were not collected during testing.	NT	NT

5. Cisco Equipment Configuration Files

Cisco 2821 Used as an IP PBX and a Router Configuration	For equipment configuration files for **Cisco 2821 used as an IP PBX and a router** go to: http://insidexo/products/converged/sip/SIP_Configruation_1.doc
Cisco 2821 Used Only as an IP PBX Configuration – with the Cisco 2811 Used Only as a Router	For equipment configuration files for **Cisco 2821 used only as an IP PBX – with the Cisco 2811 used only as a router** go to: http://insidexo/products/converged/sip/SIP_Configruation_2.doc
Cisco 2821 CUE Configuration	For equipment configuration files for **Cisco 2821 CUE** go to: http://insidexo/products/converged/sip/SIP_Configruation_3.doc
Cisco 3560 PoE Configuration	For equipment configuration files for **Cisco 3560 PoE** go to: http://insidexo/products/converged/sip/SIP_Configruation_4.doc

6. Cisco Equipment Software Versions

Cisco 2821 CME Software Version

```
Cisco2821CME#show version
Cisco IOS Software, 2800 Software (C2800NM-ADVIPSERVICESK9-M), Version
12.4(15)T, RELEASE SOFTWARE (fc3)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2007 by Cisco Systems, Inc.
Compiled Mon 25-Jun-07 22:11 by prod_rel_team

ROM: System Bootstrap, Version 12.4(13r)T, RELEASE SOFTWARE (fc1)

Cisco2821CME uptime is 1 day, 1 hour, 38 minutes
System returned to ROM by reload at 15:43:26 CDT Mon Aug 27 2007
System restarted at 15:45:21 CDT Mon Aug 27 2007
System image file is "flash:c2800nm-advipservicesk9-mz.124-15.T.bin"

This product contains cryptographic features and is subject to United States and local
country laws governing import, export, transfer, and use. Delivery of Cisco
cryptographic products does not imply third-party authority to import, export,
distribute, or use encryption. Importers, exporters, distributors, and users are
responsible for compliance with U.S. and local country laws. By using this product
you agree to comply with applicable laws and regulations. If you are unable to
comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:
http://www.cisco.com/wwl/export/crypto/tool/stqrg.html

If you require further assistance, please contact us by sending email to
export@cisco.com.

Cisco 2821 (revision 53.51) with 509952K/14336K bytes of memory.
Processor board ID FTX1103A3CF
2 Gigabit Ethernet interfaces
4 Serial interfaces
1 terminal line
4 Channelized T1/PRI ports
1 Virtual Private Network (VPN) Module
4 Voice FXS interfaces
1 Cisco service engine(s)
DRAM configuration is 64 bits wide with parity enabled.
239K bytes of non-volatile configuration memory.
125440K bytes of ATA CompactFlash (Read/Write)

Configuration register is 0x2102
Cisco2821CME#
```

6. Cisco Equipment Software Versions *(continued)*

Cisco 2821 CUE Software Version

cue# show software versions

Installed Packages	Version
Installer	2.3.2
Thirdparty	2.3.2
Bootloader (Primary)	2.1.2
Infrastructure	2.3.3
Global	2.3.2
GPL Infrastructure	2.3.0
Voice Mail	2.3.3
Bootloader (Secondary)	2.1.2
Core	2.3.0
Auto Attendant	2.3.0
Installed Languages	**Version**
English language pack	2.3.0

cue#

6. Cisco Equipment Software Versions *(continued)*

Cisco 2821 Used Only as a Router Software Version

```
Cisco2811#show version
Cisco IOS Software, 2800 Software (C2800NM-IPBASEK9-M), Version 12.4(4)T7,
RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2006 by Cisco Systems, Inc.
Compiled Tue 28-Nov-06 18:37 by kellythw

ROM: System Bootstrap, Version 12.4(13r)T, RELEASE SOFTWARE (fc1)

Cisco2811 uptime is 1 day, 1 hour, 35 minutes
System returned to ROM by reload at 20:44:00 UTC Mon Aug 27 2007
System image file is "flash:c2800nm-ipbasek9-mz.124-4.T7.bin"

This product contains cryptographic features and is subject to United States and local
country laws governing import, export, transfer, and use. Delivery of Cisco
cryptographic products does not imply third-party authority to import, export,
distribute, or use encryption. Importers, exporters, distributors, and users are
responsible for compliance with U.S. and local country laws. By using this product
you agree to comply with applicable laws and regulations. If you are unable to
comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:
http://www.cisco.com/wwl/export/crypto/tool/stqrg.html

If you require further assistance, please contact us by sending email to
export@cisco.com.

Cisco 2811 (revision 53.51) with 512000K/12288K bytes of memory.
Processor board ID FTX1102A3B9
2 FastEthernet interfaces
2 Serial interfaces
1 terminal line
2 Channelized T1/PRI ports
1 Cisco service engine(s)
DRAM configuration is 64 bits wide with parity enabled.
239K bytes of non-volatile configuration memory.
250880K bytes of ATA CompactFlash (Read/Write)

Configuration register is 0x2102
Cisco2811#
```

▶

6. Cisco Equipment Software Versions *(continued)*

Cisco Catalyst 3560 PoE Software Version

Cisco3560PoE-1#show version
Cisco IOS Software, C3560 Software (C3560-IPBASE-M), Version 12.2(25)SEE2, RELEASE SOFTWARE (fc1)
Copyright (c) 1986-2006 by Cisco Systems, Inc.
Compiled Fri 28-Jul-06 07:19 by yenanh
Image text-base: 0x00003000, data-base: 0x00EB0F14

ROM: Bootstrap program is C3560 boot loader
BOOTLDR: C3560 Boot Loader (C3560-HBOOT-M) Version 12.2(25r)SEC, RELEASE SOFTWARE (fc4)

Cisco3560PoE-1 uptime is 6 weeks, 6 days, 3 hours, 3 minutes
System returned to ROM by power-on
System image file is "flash:c3560-ipbase-mz.122-25.SEE2/c3560-ipbase-mz.122-25.SEE2.bin"

Cisco WS-C3560-24PS (PowerPC405) processor (revision P0) with 118784K/12280K bytes of memory.
Processor board ID CAT1049RJKL
Last reset from power-on
4 Virtual Ethernet interfaces
24 FastEthernet interfaces
2 Gigabit Ethernet interfaces
The password-recovery mechanism is enabled.

512K bytes of flash-simulated non-volatile configuration memory.
Base Ethernet MAC Address: 00:1A: 6D:B1:03:80
Motherboard assembly number: 73-9673-09
Power supply part number: 341-0029-05
Motherboard serial number: CAT1049516M
Power supply serial number: DTH1049C0G4
Model revision number: P0
Motherboard revision number: A0
Model number: WS-C3560-24PS-S
System serial number: CAT1049RJKL
Top Assembly Part Number: 800-25861-04
Top Assembly Revision Number: B0
Version ID: V06
CLEI Code Number: COM1X00ARC
Hardware Board Revision Number: 0x01

Switch	Ports	Model	SW Version	SW Image
* 1	26	WS-C3560-24PS	12.2(25)SEE2	C3560-IPBASE-M

Configuration register is 0xF
Cisco3560PoE-1#

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