



Oracle Enterprise Session Border Controller – Acme Packet 4600 and Microsoft Skype for Business for Enterprise SIP Trunking with CenturyLink

**Technical Application Note** 



# Disclaimer

The following is intended to outline our general product direction. It is intended for information purposes only, and may not be incorporated into any contract. It is not a commitment to deliver any material, code, or functionality, and should not be relied upon in making purchasing decisions. The development, release, and timing of any features or functionality described for Oracle's products remains at the sole discretion of Oracle.

## **Table of Contents**

INTENDED AUDIENCE	5
DOCUMENT OVERVIEW	5
INTRODUCTION	
AUDIENCE	
Requirements	6
ARCHITECTURE	6
LAB CONFIGURATION	7
PHASE 1 – CONFIGURING THE SKYPE FOR BUSINESS SERVER	8
Adding the SBC as a PSTN gateway	
CREATING A ROUTE WITHIN THE LYNC SERVER INFRASTRUCTURE	
Additional Steps	
PHASE 2 – CONFIGURING THE ORACLE ENTERPRISE SBC	23
IN SCOPE	
OUT OF SCOPE	23
WHAT WILL YOU NEED	23
SBC- GETTING STARTED	
Establish the serial connection and logging in the SBC	
Initial Configuration – Assigning the management Interface an IP address	
CONFIGURING THE SBC	
SIP PRACK interworking	
Local REFER handling by the SBC	
Ring-back tone during Transfers	
SIP manipulations	
SBC CONFIGURATIONS	
SBC CONFIGURATION FOR SINGLE TRUNK REGISTRATION USING PCMU TOWARDS TRUNK	
SBC CONFIGURATION FOR SINGLE TRUNK REGISTRATION USING G729 TOWARDS TRUNK	
TEST PLAN EXECUTED	
CONFIGURATION FOR DUAL TRUNK REGISTRATION AND TEST PLAN	67
SBC CONFIGURATION FOR DUAL TRUNK REGISTRATION USING PCMU TOWARDS TRUNK	67
SBC CONFIGURATION FOR DUAL TRUNK REGISTRATION USING G729 TOWARDS TRUNK	
TEST PLAN	
TROUBLESHOOTING TOOLS	103
MICROSOFT NETWORK MONITOR (NETMON)	
WIRESHARK	
Eventviewer	
On the Oracle SBC 4600 Series	
Resetting the statistical counters, enabling logging and restarting the log files	
Examining the log files	
Through the Web GUI	
Telnet	
On the Lync Server	
Lync Server Logging Tool	
APPENDIX A	106

# ACCESSING THE ACLI106ACLI BASICS106CONFIGURATION ELEMENTS108CREATING AN ELEMENT108EDITING AN ELEMENT109DELETING AN ELEMENT109CONFIGURATION VERSIONS109SAVING THE CONFIGURATION110

## **Intended Audience**

This document is intended for use by Oracle personnel, third party Systems Integrators, and end users of the Oracle Enterprise Session Border Controller (E-SBC). It assumes that the reader is familiar with basic operations of the Oracle Enterprise Session Border Controller – Acme Packet 4600.

## **Document Overview**

Microsoft Skype for Business offers the ability to connect to SIP based telephony trunks using an IP communication. This reduces the cost and complexity of extending an enterprise's telephony system outside its network borders. Oracle Enterprise Session Border Controllers (E-SBCs) play an important role in SIP trunking as they are used by many trunk providers and some enterprises as part of their SIP trunking infrastructure.

This application note has been prepared as a means of ensuring that SIP trunking between Microsoft Skype for Business servers, Oracle E-SBCs and IP Trunking services are configured in the optimal manner.

It should be noted that the E-SBC configuration provided in this guide focuses strictly on the Skype for Business Server associated parameters. Many E-SBC users may have additional configuration requirements that are specific to other applications. These configuration items are not covered in this guide. Please contact your Oracle representative with any questions pertaining to this topic.

For additional information on Skype for Business Server, please visit http://www.skype.com/en/business/.

## Introduction

#### Audience

This is a technical document intended for telecommunications engineers with the purpose of configuring the Oracle Enterprise SBC and the Skype for Business Server. There will be steps that require navigating Microsoft Windows Server as well as the Acme Packet Command Line Interface (ACLI). Understanding the basic concepts of TCP/UDP, IP/Routing, and SIP/RTP are also necessary to complete the configuration and for troubleshooting, if necessary.

#### Requirements

- Fully functioning Skype for Business Server deployment, including Active Directory and DNS
- A dedicated Mediation Server for the SIP trunking connection
- Microsoft Skype for Business 2015 Version 6.0.93190.0
- Skype for Business 2015 client, Version 15.0.4753.1000
- Oracle Enterprise Session Border Controller AP 4600 running Net-Net OS ECZ730m1p1.64.bz. Note: the configuration running on the SBC is backward/forward compatible with any release in the 7.3.0 stream.

#### Architecture

The following reference architecture shows a logical view of the connectivity between Skype for Business Server and the SBC.



The Enterprise Network represents the customer's on-premise infrastructure, which includes the Active Directory, DNS and Skype for Business Server systems. The SIP Trunk Network represents the infrastructure of the SIP trunk provider, CenturyLink, which provides PSTN service via the SIP trunk. Area C represents the integration of these two environments over an IP network. This could be, through a VPN tunnel over the Internet, an MPLS managed network, or even a dedicated physical connection. The SFB Mediation Server and the SBC are the edge components that form the boundary of the SIP trunk.

As per the CenturyLink network requirements, the customer devices are required to register and support authentication. The Oracle ESBC supports authentication and performs registration on behalf of the Skype for Business Server. The registration can be done using either the single trunk architecture or the dual trunk architecture. In the single trunk architecture, the Oracle ESBC registers a single trunk to the CTL Carrier SBC and in the dual trunk architecture; the ESBC registers to a pair of geo-redundant CTL carriers SBC. This guide documents both single and dual trunk architectures, starting with the configuration required for the testing of the single trunk architecture.

The configuration, validation and troubleshooting of the areas B and C is the focus of this document and will be described in two phases:

- Phase 1 Configure the Skype for Business Server
- Phase 2 Configure the 4600

#### Lab Configuration

The following diagram, similar to the Reference Architecture described earlier in this document, illustrates the lab environment created to facilitate certification testing (IP addressing/Port below is only a reference, they can change per your network specification).



## Phase 1 – Configuring the Skype for Business server

The enterprise will have a fully functioning Skype for Business Server infrastructure with Enterprise Voice deployed and a Mediation Server dedicated to this installation. If there is no Mediation Server present for this purpose, one will have to be deployed.

There are two parts for configuring SFB Server to operate with the Oracle SBC:

- Adding the SBC as a PSTN gateway to the SFB Server infrastructure
- Creating a route within the SFB Server infrastructure to utilize the SIP trunk connected through the SBC.

To add the PSTN gateway, we will need:

- IP addresses of the external facing NICs of the Mediation Servers
- IP address of the sip interface of the SBC facing the Mediation servers
- Rights to administer Lync Server Topology Builder
- Access to the Lync Server Topology Builder

Note: This section of the Application note only walks you through adding Oracle E-SBC to Microsoft's Skype for Business config, The assumption is that Microsoft's Skype for Business application is already installed and 100% functional.

#### Adding the SBC as a PSTN gateway

The following process details the steps to add the SBC as the PSTN gateway

- 1. On the server where the Topology Builder is located, start the console.
- 2. From the Start bar, select Lync Server Topology Builder.



3. The opening screen of the Topology builder will be displayed. Click on the **Cancel** button.



4. The Topology Builder window will now be displayed. Click on Action and select Download Topology.

16			Skype for Business Server 2015, Topology Builder	_ <b>D</b> ×
File	Actio	n Help		
		New Central Site	Define a new deployment from the Actions pane	
		Edit Properties		
		New Topology		
		Open Topology		
	4	Download Topology		
		Save a copy of Topology As	Download complete deployment topology from the Central Management sto	re.
		Publish Topology		
		Install or upgrade a database.		
	-	Kemove Deployment		
		Help		
				Activate Wir

5. You will then see a screen showing that you have successfully imported the topology. Click the **Ok** button.

20	Download Topology
Downlo	ading topology
Succeed	fed
Downlo	ading global simple URL settings
Succeed	fed
Finished	1
	OK Cancel
-	

6. Next you will be prompted to save the topology which you have imported. You should revision the name or number of the topology according to the standards used within the enterprise. Click the **Save** button

Note: This keeps track of topology changes and, if desired, will allow you to fall back from any changes you make during this installation

0	Save	e Topology As		×
🔄 🔄 👻 🕇 🎴	« Documents + temp	~ ¢	Search temp	Q.
Organize 👻 Ne	w folder		8== -	
☆ Favorites	Name	•	Date modified	Type More opt
Desktop  Downloads  Recent places	=	No items match y	your search.	
Documents				
<ul> <li>Pictures</li> <li>Videos</li> </ul>				
🛤 Computer	~ <	ш		
File <u>n</u> ame:	Current2			
Save as type:	Topology Builder files (*.tb	xml)		
Hide Folders			Save Car	icel

7. You will now see the topology builder screen with the enterprise's topology imported.

15	Skype for Business Server	r 2015, Topology Builder 🛛 📃 🗖 🗙
File Action Help		
Skype for Business Server	SIP domain	•
r 🔛 Cleanderant Optidgy	Default SIP domain: Additional supported SIP domains:	sfblabdm.local Not configured
	Simple URLs	<b>^</b>
	Phone access URLs:	Active Simple URL V https://dialins.fblabdm.local
	Meeting URLs:	Active Simple URL SIP domain
	Administrative access URL:	https://admin.sfblabdm.local
	Central Management Serv	rver 🔺
	Central Management Server:	Active         Front End         Site           ✓         fe01.sfblabdm.local         CleanDefaultTop
		Activota Mindowe

8. In the upper left hand corner, expand the site in which the PSTN gateway will be added. In our case, the site is **Bedford**. Then click on the **PSTN Gateways** 



9. Right click on PSTN gateways and select New IP/PSTN Gateway.

200 - Contra 1997 - Contra 199	Skype for Business Server 2015, Topology Builder	_ <b>D</b> ×
File Action Help		
File     Action     Help <ul> <li>             GenDefaultTopology         </li> </ul> <ul> <li>             GenDefaultTopology         </li> <li>             GenDefaultTopology         </li> <li>             Generation</li> <li>             Skype for Business Server 2015         </li> <li>             Shared Components         </li> <li>             Sile Stores         </li> <li>             Set Stores         </li> </ul> <ul> <li>             Set Stores         </li> </ul> File stores                 Office Web App                 SIP Video gateways                 SIP Video trunks                Branch sites	The properties for this item are not available for editing.  Sateway Define a new IP/PSTN gateway.	

10. In the **Define New IP/PSTN Gateway** window, enter the ip address of the SIP interface of the SBC in the **FQDN** text box and click **Next**.

10	Define New IP/PSTN Gateway	×
5	Define the PSTN Gateway FQDN	
Define ti FQDN: *	he fully qualified domain name (FQDN) for the PSTN gateway.	
oracle.s	sfblabdm.local	
Help	Back Next Cancel	]

11. Select Enable IPv4 in the Define the IP address section and click Next.

9	Define New IP/PSTN Gateway	x
5	Define the IP address	
● Enat ● L ○ L	ole IPv4 Use all configured IP addresses. Limit service usage to selected IP addresses. PSTN IP address:	
O Enat ⊚ t ⊖ t	ole IPv6 Use all configured IP addresses. Limit service usage to selected IP addresses. PSTN IP address:	
Help	Back Next Cancel	

12. In the next section, enter the ip address of the SBC's sip interface under **Trunk name**. Configure the **Listening port for IP/PSTN gateway** as 5068, TCP as the **SIP Transport Protocol** and click **Finish**.

Define New IP/PSTN Gateway	2
Define the root trunk	
Trunk name: *	
oracle.sfblabdm.local	
Listening port for IP/PSTN gateway: *	
5066	
SIP Transport Protocol:	
TCP 🗸	
Associated Mediation Server:	
medpool.sfblabdm.local CleanDefaultTopology	
Associated Mediation Server port: *	
5060	
Help Back Finish Cancel	

13. The PSTN gateway for the Lync server has been added. It will be listed under **PSTN gateways**.

Expand the **Mediation Pool** list and click on the Mediation Server to be utilized. In our example the Mediation Server is **sfbmedpool.acmepacket.net**.

	Skype for E	Business Server 201	5, Topology Buil	der		×
File Action Help						
<ul> <li>▲ B Skype for Business Server</li> <li>▲ B CleanDefaultTopology</li> </ul>	General					
<ul> <li>Lync Server 2010</li> <li>Lync Server 2013</li> </ul>	FQDN:	medpool.sfblabdm.lo	cal			
A Description of the second	Associations					
Standard Edition Front End Servers	Edge pool (for media):	Not associated				
Director pools	Note: To view the federation	on route, use the site pr	operty page.			
Mediation pools						
Ę <sub>Q</sub> fe01.sfblabdm.local	Next hop selection				•	
Wey medpool.stblabdm.local     P ugy medpool.stblabdm.local     Persistent Chat pools     Edge pools     Trusted application servers	Next hop pool:	<u>fe01.sfblabdm.local</u>	CleanDefaultTopolog	avì		
Video Interop Server pools A Discrete Components	Mediation Server PSTN gat	eway			•	
<ul> <li>SQL Server stores</li> <li>SQL Server stores</li> <li>File stores</li> </ul>	TLS listening port: TCP listening port:	5067 - 5067 5060 - 5060				
PSTN gateways	Trunks:	Default	Trunk	Gateway	Site	1
Grundessibilabulin.iocai     Trunks     Go Office Web Apps Servers		<u>oracle.sfbla</u>	abdm.local	oracle.sfblabdm.local	<u>CleanDefaultTopology</u>	
🔁 Video gateways						
🚞 SIP Video trunks						
🚞 Branch sites						
				Activate Min	dows	

You will see that the PSTN gateway is associated with the Mediation server.

14. In the upper right hand corner of your screen under Actions select Topology then select Publish.

Skype for Business Server 2015, Topology Builder						x
File Action Help						
⊿ Resserver ∠ Edit Properties	General					<u> </u>
Topology Delete Help Enterprise Edition Fr Director pools	oad Current Topology Copy	medpool.sfblabc Not associated tion route, use the si	im.local te property page.			
Director pools     Install o breade a database     Remove Deployment     Remove Deployment     Persistent Chat pools     Persistent Chat pools     Next hop pool:			cal (CleanDefaultTopology)			<u> </u>
Trusted application servers Video Interop Server pools Mediation Server PSTN g Mediation Server PSTN g		gateway				<b>_</b>
GQL Server stores     TLS listening port:     File stores     TCP listening port:     PSTN gateways		5067 - 5067 5060 - 5060				
<ul> <li>oracle.sfblabdm.local</li> <li>Trunks</li> <li>Office Web Apps Servers</li> <li>Video gateways</li> <li>SIP Video trunks</li> <li>Branch sites</li> </ul>	I runks:	Default <u>oracle</u>	Trunk stblabdm.local	Gateway oracle.sfblabdm.local	Site CleanDefaultTopology	

You will now see the Publish Topology window. Click on the Next button

10	Publish Topology	×
Publis	h the topology	
In order publish y complet	for Skype for Business Server 2015 to correctly route messages in your deployment, you must your topology. Before you publish the topology, ensure that the following tasks have been ed:	
• A va	lidation check on the root node did not return any errors.	^
• A fil • All s	e share has been created for all file stores that you have configured in this topology. imple URLs have been defined.	
For Arch exce	Enterprise Edition Front End pools and Persistent Chat pools and for Monitoring Servers and iving Servers: All SQL Server stores are installed and accessible remotely, and firewall eptions for remote access to SQL Server are configured.	=
For com	a single Standard Edition server, the "Prepare first Standard Edition server" task was apleted.	
<ul> <li>You sysa</li> </ul>	are currently logged on as a SQL Server administrator (for example, as a member of the SQL admin role).	
If yo     When yo	bu are removing a Front End pool, all users, common area phones, analog devices, application tast objects, and conference directories have been removed from the pool us are ready to proceed, click Next.	~
Help	Back Next Cancel	_

You will now be at a window showing the databases associated with site. Click Next.

9	Publish Topolo	gy	x
Put	lishing wizard complete		
Your	topology was successfully published.		
****	Step Publishing topology Downloading topology Downloading global simple URL settings Updating role-based access control (RBAC) roles Enabling topology	Status Success Success Success Success Success	View Logs
To cl	ose the wizard, click Finish. Ielp	Back Finish	Cancel

- 15. When complete you should see a window from Topology Builder stating that your topology was successfully published. Click the **Finish** button.
- 16. You will be at the Topology Builder main window, expand your site and double check that your PSTN entries are correct and that the appropriate Mediation Server has the PSTN gateway associated.

#### Creating a route within the Lync Server infrastructure

In order for the Lync Server Enterprise Voice clients to utilize the SIP trunking infrastructure that has been put in place, a route will need to be created to allow direction to this egress. Routes specify how Lync Server handles calls placed by enterprise voice users. When a user places a call, the server, if necessary, normalizes the phone number to the E.164 format and then attempts to match that phone number to a SIP Uniform Resource Identifier (URI). If the server is unable to make a match, it applies outgoing call routing logic based on the number. That logic is defined in the form of a separate voice route for each set of target phone numbers listed in the location profile for a locale. For this document we are only describing how to set up a route. Other aspects which apply to Lync Server Enterprise Voice deployments such as dial plans, voice policies, and PSTN usages are not covered.

To add the route we will need:

- Rights to administer Lync Server Control Panel
  - Membership in the CS Administrator Active Directory Group
- Access to the Lync Server Control Panel

The following process details the steps to create the route:

1. From the Start bar, select Lync Server Control Panel.



You will be prompted for credential, enter your domain username and password.

2. Once logged in, you will now be at the "Welcome Screen". On the left hand side of the window, click on Voice Routing.



3. The Dial Plan tab in the Voice Routing section will be displayed. On the content area toolbar, click +New.

Skype for Busi	Skype for Business Server 2015 Control Panel	Administrator   Sign out 6.0.9319.0   Privacy statement
Home	DIAL PLAN VOICE POLICY ROUTE PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING	
Users Topology	Create voice routing test case information	~
IM and Presence Persistent Chat	٩	
Voice Routing	🗣 New 🔻 🧪 Edit 🔻 Action 🔻 Commit 💌	0
Voice Features Response Groups	Name  Scope State Normalization rules Description	
Conferencing		
Federation and External Access		
Monitoring and Archiving		
Security		
Network Configuration		

4. Next you build a Dial Plan and a translation rule for the phone numbers you want this route to handle.

US Dial-plan

Match this pattern: ^(\d\*)\$

Translation rule: +1\$1

Additional normalization rules will need to be created as shown below to allow dialing the feature codes.



5. On the top row of the tabs, select **Route**. On the content area toolbar, click **+New**.

<b>F</b> 6	Skype for B	usiness Server 20	15 Control Panel		_ 🗆 🗙
Skype for Busine	ss Server				Administrator   Sign out 5.0.9319.0   Privacy statement
Home Users Topology	DIAL PLAN VOICE POLICY ROU	TE PSTN USAGE	TRUNK CONFIGURATION	TEST VOICE ROUTING	~
IM and Presence Persistent Chat			Q		
Voice Routing Voice Features	♣ New Edit ▼ Move up Name	Move down	Action ▼ Commit ▼	Pattern to match	0
Response Groups Conferencing Clients Federation and External Access Monitoring and Archiving Security Network Configuration					

6. On the **New Voice Route** page, in the **Name** field, enter the name you have selected for the Route. In our example, it is US route.

5	Skype for Business Server 2015 Control Panel	_ <b>D</b> X
Skype for Busine	is Server	Administrator   Sign out 6.0.9319.0   Privacy statement
Home	DIAL PLAN VOICE POLICY ROUTE PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING	
Users Topology	Create voice routing test case information	~
IM and Presence	New Voice Pouto	
Persistent Chat		Ø
Voice Routing		<u>م</u>
Voice Features	Name: *	
Response Groups		
Conferencing	Description:	
Clients		
Federation and External Access	Build a Pattern to Match Add the starting digits that you want this route to handle, or create the expression manually by clicking Edit.	
Monitoring	Starting digits for numbers that you want to allow:	
and Archiving	Type a valid number and then click Add. Add	
Security	Exceptions	
Configuration	Remove	
	Match this pattern: *	

7. Next you build a Pattern Match for the phone numbers you want this route to handle. Click Edit.

Skype for Business Server	Administrator   Sign ou 6.0.9319.0   Privacy statemer
Home         Users         Topology         IM and Presence         Persistent Chat         Voice Routing         Voice Routing         Voice Routing         Voice Features         Response Groups         Conferencing         Clients         Clients         Monitoring and Archiving         Security         Network Configuration	TRUNK CONFIGURATION TEST VOICE ROUTING

8. Next you want to associate the Voice Route with the **Trunk** you have just created. Scroll down to **Associated Trunks**, click on the **Add** button.

√ OK X Cancel		
Add the starting digits that you want this route to ha the expression manually by clicking Edit. Starting digits for numbers that you want to allow:	indle, or create	
Type a valid number and then click Add.	Add	
	Exceptions	
	Remove	
Match this pattern: *		
* *		
Edit Reset ?		
Suppress caller ID		
Alternate caller ID:		

9. You will now be at a window showing available Trunks to associate your Voice Route. Click on the PSTN gateway that you just created and then click the **OK** button.

elec	ct Trunk		2	
			9	
	Service	Site		
	PstnGateway:oracle.sfblabdm.local	CleanDefaultTopology		
		OK	Cancel	

10. You can now see that you have associated your trunk with the route you created. An appropriate PSTN usage record will need to be assigned as well. In our example, we use one that was already created in the enterprise. Click on the **Select** button under **Associated PSTN Usages**.

New Voice Route	
ok 🗙 Cancel	•
Associated trunks:	•
PstnGateway:oracle.sfblabdm.local	Add
	Remove
Associated PSTN Usages	
🞦 Select Remove 👚 🐥	
PSTN usage record Associated vo	e policies
	-

11. In the Select PSTN Usage Record window displayed, select the appropriate PSTN Usage Record and click OK.

Select PSTN Usage Record			2	23
		2		
PSTN usage record name	Associated routes	Associated voice policies		
Internal		Global		
Local		Global		
Long Distance	ECB	Global		
		ОК	ancel	

12. You will now see the Associated PSTN Usages which you have added. Click the **OK** button at the top of the **New Voice Route** screen.

New	Voice	Route

Associated trunks: Associated trunks: PstnGateway:oracie sfblabdm.local Remove Remove Select Remove PSTN Usage record Long Distance Global	JOK X Cancel		
Associated trunks: PstnGateway:oraciesfblabdm.local Add Remove Sociated PSTN Usages Select Remove  Select Remove  Sociated voice policies Long Distance Global			
PstnGateway:oracie sfblabdm.local Add Remove	Associated trunks:		
Remove      Resociated PSTN Usages	PstnGateway:oracle.sfblabdm.local	Add	
Associated PSTN Usages Select Remove  Select		Remove	
Associated PSTN Usages			
Associated PSTN Usages   Select   Remove   Select   Remove   Associated voice policies   Long Distance   Global			
Select     Remove     Image: Composition of the com	Associated PSTN Usages		
PSTN usage record     Associated voice policies       Long Distance     Global	🛀 Select Remove 👚 🦊		
Long Distance Global	PSTN usage record Associated voice police	ies	
	Long Distance Global		

13. You will now be at the Routes page showing the US route. Click the Commit drop-down menu, and then Commit All.

Skype for Busi	ness Server	Administrator   Sign out 6.0.9319.0   Privacy statement
Home	DIAL PLAN VOICE POLICY ROUTE PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING	
Topology	Create voice routing test case information	~
IM and Presence		
Persistent Chat	Q	
Voice Routing		0
Voice Features	New Zelit      Those up Move down Action      Commit      Commit      Nove a second action      Commit      Review uncommitted changes when	U
Response Groups	US route Uncommitted Long Distar Commit all [10])\$	
Conferencing	LocalRoute PSTN_Usag Cancel selected changes	
Clients	ECB Uncommitted Long Distar Cancel all uncommitted changes 10))\$	
Federation and External Access		
Monitoring and Archiving		
Security		
Network Configuration		

14. On the Uncommitted Voice Configuration Settings window, click OK.

Uncommitted Voice Configuration Settings					(?)	23
R	outes				^	
	Identity	Action	New value (pattern to match)	Old value (pattern to match)		
	US route	Modified	^(\+1[0-9]{10})\$	^(\+1[0-9]{10})\$		
	LocalRoute	Modified	*	.*		
					ancei	

If there are no errors, the new Voice Route has now been successfully created and the state will show as Committed.

**Additional Steps** 

There are other aspects to a Lync Server Enterprise Voice deployment such as

- Site, local, and global dial plans
- Voice Policies
- Assigning Voice Policies to users
- PSTN usage policies

Refer to MSDN technet for relevant information.

## Phase 2 – Configuring the Oracle Enterprise SBC

In this section we describe the steps for configuring an Oracle Enterprise SBC, formally known as an Acme Packet Net-Net Session Director ("SBC"), for use with Skype for Business Server in a SIP trunking scenario.

#### In Scope

The following guide configuring the Oracle SBC assumes that this is a newly deployed device dedicated to a single customer. If a service provider currently has the SBC deployed and is adding SFB Server customers, then please see the ACLI Configuration Guide on <a href="http://docs.oracle.com/cd/E61547\_01/index.html">http://docs.oracle.com/cd/E61547\_01/index.html</a> for a better understanding of the Command Line Interface (CLI).

Note that Oracle offers several models of SBC. This document covers the setup for the 4600 platform series running Net-Net OS ECZ7.3.0 or later. If instructions are needed for other Oracle SBC models, please contact your Oracle representative.

#### Out of Scope

Configuration of Network management including SNMP and RADIUS; and

#### What will you need

- Serial Console cross over cable with RJ-45 connector
- Terminal emulation application such as PuTTY or HyperTerm
- Passwords for the User and Superuser modes on the Oracle SBC
- IP address to be assigned to management interface (Wancom0) of the SBC the Wancom0 management interface must be connected and configured to a management network separate from the service interfaces. Otherwise the SBC is subject to ARP overlap issues, loss of system access when the network is down, and compromising DDoS protection. Oracle does not support SBC configurations with management and media/service interfaces on the same subnet.
- IP address of Mediation Server external facing NIC
- IP addresses to be used for the SBC internal and external facing ports (Service Interfaces)
- IP address of the next hop gateway in the service provider network
- IP address of the enterprise DNS server

#### **SBC- Getting Started**

Once the Oracle SBC is racked and the power cable connected, you are ready to set up physical network connectivity. **Note: use the console port on the front of the SBC, not the one on the back.** 



Plug the slot 0 port 0 (s0p0) interface into your outside (gateway facing) network and the slot 0 port 1 (s1p0) interface into your inside (SFB server-facing) network. Once connected, perform you are ready to power on and perform the following steps.

All commands are in bold, such as **configure terminal**; parameters in bold red such as **oraclesbc1** are parameters which are specific to an individual deployment. **Note:** The ACLI is case sensitive.

Establish the serial connection and logging in the SBC

Confirm the SBC is powered off and connect one end of a straight-through Ethernet cable to the front console port (which is active by default) on the SBC and the other end to console adapter that ships with the SBC, connect the console adapter (a DB-9 adapter) to the DB-9 port on a workstation, running a terminal emulator application such as PuTTY. Start the terminal emulation application using the following settings:

- Baud Rate=115200
- Data Bits=8
- Parity=None
- Stop Bits=1
- Flow Control=None

Power on the SBC and confirm that you see the following output from the bootup sequence.

🚱 COM3 - PuTTY 🔄 💽 💽 💌
Starting tEbmd
Starting tSipd
Starting tLrtd
Starting tH323d
Starting tH248d
Starting tBgfd
Starting tSecured
Starting tAuthd
Starting tCertd
Starting tIked
Starting tauditd
Starting tauditpusher
Starting tSnmpd
Start platform alarm
Initializing /ramdrv Cleaner
Starting tLogCleaner task
Bringing up shell
password secure mode is enabled
Admin Security is disabled
Starting SSH
SSH_Cli_init: allocated memory for 5 connections
acli: max telnet sessions: 5
Password: 0x21a059c8 (tAlarm): eth0: Link is up (1000Mb/s full duplex)

Enter the following commands to login to the SBC and move to the configuration mode. Note that the default SBC password is "**acme**" and the default super user password is "**packet**".

Password: acme oraclesbc1> enable Password: packet oraclesbc1# configure terminal oraclesbc1(configure)#

You are now in the global configuration mode.

Initial Configuration - Assigning the management Interface an IP address

To assign an IP address, one has to configure the bootparams on the SBC by going to

oraclesbc1#configure terminal --- >bootparams

- Once you type "bootparam" you have to use "carriage return" key to navigate down
- A reboot is required if changes are made to the existing bootparams

```
ACMESYSTEM(configure) # bootparam

'.' = clear field; '-' = go to previous field; q = quit

Boot File : /boot/nnECZ730m1p1.64.bz

IP Address : 192.65.79.44

VLAN :

Netmask : 255.255.224
```

Gateway	: 192.65.79.33
IPv6 Address	:
IPv6 Gateway	:
Host IP	: 0.0.0.0
FTP username	: vxftp
FTP password	: vxftp123
Flags	:
Target Name	: ACMESYSTEM
Console Device	: COM1
Console Baudrate	: 115200
Other	:
NOTE: These changed p	parameters will not go into effect until reboot.
Also, be aware that a	some boot parameters may also be changed through
PHY and Network Inte:	rface Configurations.

#### Configuring the SBC

The following section walks you through configuring the Oracle Enterprise SBC required to work with Skype for Business (SFB) and CenturyLink SIP trunk. The information below is split into three sections:

- The single trunk test plan utilizing the G711 codec The transport protocol for signaling/media between SBC and SFB is TCP and RTP – the SIP trunk utilizes UDP/RTP.
- The single trunk test plan utilizing the G729 codec The transport protocol for signaling/media between SBC and SFB is TCP and RTP using the G711 codec – the SIP trunk utilizes UDP/RTP and supports G729 codec, the SBC performs transcoding in this scenario.

It is outside the scope of this document to include all the interoperability working information as it will differ in every deployment.

#### SIP PRACK interworking

In order to establish an early media session for outbound calls, Skype for Business gateway specification mandates the PSTN gateways to offer a reliable provisional response and for inbound calls offer INVITEs with a supported header. The SBC interworks the messaging and provide RFC 3262 PRACK interworking towards SFB and it is a mandatory configuration in all Oracle ESBC–Microsoft SFB deployments. The following need to be configured on ESBC

- Configure option 100rel-interworking on the sip-interface facing mediation server
- Configure a sip-feature to pass the 100rel in supported and require headers
- Configure a sip-manipulation (ForEarlyMedia) to add a Require:100rel header in incoming SIP INVITE from mediation server and delete the Supported:100rel header

#### Local REFER handling by the SBC

In event of transferring calls from a SFB client (calls could be initially placed between 2 SFB Clients, but then transferred out to PSTN), SFB has two OPTIONS:

- Transfer using re-INVITE approach
- Transfer using REFER method --- the info below is required when REFER based transfer is enabled.

Oracle E-SBC supports REFER method termination when received from SFB. After termination the ESBC uses the info in refer-to header and sends a newly generated INVITE towards the SFB Mediation server. SFB then processes the INVITE, authorizes the call transfer and sends a new INVITE (for calls transferred to PSTN) to the SBC or transfers call internally to the transferred SFB client

To handle the call transfer and refer scenarios – when SFB client 1 refers/transfers the call to SFB Client 2 or to a party on the PSTN, we will need two routes to route to the two mediation servers depending on the referred party:

local-policy		
from-address	*	
to-address	sfbmedpool.acmepacket.net	
source-realm	SIP-Trunk	
description	For referred party header	
activate-time		
deactivate-time		
state	enabled	
policy-priority	none	
policy-attribute		
next-hop	sfbmedpool.acmepacket.net	
realm	towards-sfb	
action	replace-uri	
terminate-recursion	disabled	
carrier		
start-time	0000	
end-time	2400	
days-of-week	U-S	
cost	0	
state	enabled	
app-protocol	SIP	
methods		
media-profiles		
lookup	single	
next-key		
eloc-str-lkup	disabled	
eloc-str-match		

**Ring-back tone during Transfers** 

During call transfer to a PSTN party, the transfer completes but the calling party does not hear a ring back tone during transfer. The INVITE Lync sends to the SBC to initiate the transfer contains the SDP attribute, a=inactive which is forwarded to the trunk and as a result of which the SBC cannot play the ring back tone to the original PSTN caller (while call is being transferred). A sendonly attribute is required for MoH and transfer scenarios for the calling party to be able to hear ringback or MoH when it is kept on hold. The SBC is able to signal appropriately towards the SIP trunk by changing the "a=inactive" SDP attribute in the INVITE to sendonly towards PSTN.

Sip manipulations are configured to make the necessary changes. The manipulation ForEarlyMedia is configured to change the SDP attribute from a=inactive to a=sendonly in the INVITEs sent to the calling party for transfer

sip-manipulation		
name	Changeinactosendonly	
description	Change inactive to sendonly for transfer	
split-headers		
join-headers		
header-rule		
name	changeSDP	
header-name	Content-Type	
action	manipulate	
comparison-type	case-sensitive	
msg-type	request	
methods	INVITE	
match-value		
new-value		
element-rule		
name	inacttosendonly	
parameter-name	application/sdp	

type	mime
action	find-replace-all
match-val-type	any
comparison-type	pattern-rule
match-value	a=inactive
new-value	a=sendonly

We utilize the local playback feature of the SBC to play ring back tone during transfers. The ringback tone is played based on REFER termination. You must upload an audio file (in .raw format) to /code/media onto the ESBC for the media you want played during the transfer. A separate file is required for each different codec type, even if the media itself is the same.

The playback configuration is defined listing the media files that you want to play. The playback-config element is configured under media-manager.

playback-config	
name	transferrbt
entry	
encoding	PCMU
filename	US ringbackPCMU.raw
bytes-per-sec	8000

The playback options can be applied to realms, sip-interfaces or session agents using the spl-options command.

```
oraclesbc1(session-router)# sip-interface
oraclesbc1(sip-interface)# sel
<realm-id>:
1: towards-sfb 192.168.2.225:5060
2: SIP-trunk 192.168.1.220:5060
selection: 1
oraclesbc1(sip-interface)# spl-options playback-on-refer="transferrbt"
oraclesbc1(sip-interface)# done
```

#### **SIP** manipulations

SFB and SIP trunk carry their own SIP and SDP design – not always these implementation methods align, causing a lot of mismatch in SIP and SDP signaling and call flow. The ESBC helps resolve these issues with SIP manipulation feature. In this section, we detail the changes that we implement to make sure the signaling is per the specifications of the trunk and SBC.

The HMR applied to the signaling towards the trunk performs the following changes:

- The Request-URI is modified to include the ip address and port of the trunk device
- The P-Asserted-Identity header sent by SFB is deleted and a new PAI header is added to include the pilot number that is used to register with the trunk, in our case we delete the original PAI header and add a PAI with the value sip:4695550016@voip.centurylink.com
- The uri-host portion of the From header is replaced with the FQDN of the trunk, in our case the uri-host is changed to voip.centirylink.com
- In the Contact header, we have header rules to strip +1 from the uri-user and replace the uri-host and uri-port portions with the SBC's local ip and port of the interface facing the trunk.
- In the Route header we remove the +1 from the uri-user.
- For privacy enabled calls, SFB sends the phone number in the From header. It indicates that it is a privacy enabled calls using the 'Privacy:id' header. For such calls, we replace the phone number in the uri-user of the From header with 'anonymous'.

To conform SFB's signaling per the trunk's specification, we modify the messages coming from SFB and also make some changes to messages before they are sent to SFB.

The following changes are applied to the messages coming from SFB:

- We add a 'Require:100rel' header in incoming SIP INVITE from mediation server and delete the 'Supported:100rel' header as mentioned in the SIP PRACK interworking section.
- To enabled ringback on transfers, we replace the 'a=inactive' line in SDP of the INVITEs with 'a=sendonly'. For more information, please refer to the Ring-back tone during Transfers section.

To the messages sent to SFB, the following changes are applied:

- The uri-hosts of the From and To headers are replaced with SBC's local ip and SFB's ip.
- In the From and To headers we remove the +1 from the uri-user, when the uri-user is anonymous.
- At last we have a rule to insert +1 in the uri-user of the Contact header as SFB server is configured for E.164 format.

## SBC Configurations

#### SBC configuration for Single Trunk registration using PCMU towards trunk

Following is the configuration of the SBC:

local-p	olicy		
	from-address	*	
	to-address	*	
	source-realm	core	
	policy-attribute		
	next-hop		192.168.1.220
	realm		ctl-trunk
local-p	olicy		
	from-address	*	
	to-address	SFBMed1	.partnersfb.com
	source-realm	core	
	policy-attribute		
	next-hop		SFBMed1.partnersfb.com
	realm		core
	action		replace-uri
local-p	olicy		
	from-address	*	
	to-address	SFBMed2	.partnersfb.com
	source-realm	core	
	policy-attribute		
	next-hop		SFBMed2.partnersfb.com
	realm		core
	action		replace-uri
local-p	olicy		
	from-address	4695550	016
	to-address	*	
	source-realm	core	
	policy-attribute		
	next-hop		192.168.1.220
	realm		ctl-trunk
local-p	olicy		
	from-address	*	
	to-address	*	

source-realm	ctl-trunk
policy-attribute	
next-hop	medpool.partnersfb.com
realm	core
media-manager	
media-policy	
name	voip-default.
tos-settings	
media-type	audio
tos-value	0x28
tos-settings	
media-type	message
media-sub-type	sin
tos-value	0x18
network-interface	0410
name	slool
in-address	192 168 1 220
netmask	255 255 255 0
actoway	102 168 1 1
hin-in-list	102 160 1 220
icmp-addross	192.168.1.220
network-interface	192.100.1.220
name	e1p0
hostnamo	5100
in-addross	102 168 / 135
netmask	255 255 255 0
ateway	192 168 4 1
dns-in-primary	192.168 / 150
dns-domain	nartnersfb.com
hin-in-list	192 168 4 135
icmp-address	192 168 4 135
nhy-interface	192.100.1.100
name	slini
operation-type	Media
phy-interface	
name	s1p0
operation-type	Media
slot	1
playback-config	
name	transferrbt
entry	
encoding	PCMU
filename	US ringbackPCMU.raw
realm-config	
identifier	core
network-interfaces	s1p0:0
mm-in-realm	enabled
qos-enable	enabled
restricted-latching	sdp
realm-config	
identifier	ctl-trunk
network-interfaces	s0p0:0
mm-in-realm	enabled
qos-enable	enabled
media-policy	voip-default
session-agent	

hostname ip-address port realm-id out-translationid out-manipulationid session-agent hostname port transport-method realm-id ping-method ping-interval out-translationid in-manipulationid out-manipulationid refer-call-transfer auth-attributes auth-realm username password in-dialog-methods session-translation id rules-calling rules-called session-translation id rules-calling rules-called sip-config home-realm-id registrar-domain registrar-host registrar-port options sip-message-len refer-src-routing sip-feature name realm require-mode-inbound require-mode-outbound sip-interface realm-id sip-port address transport-protocol allow-anonymous registration-caching options spl-options sip-interface realm-id sip-port

192.168.1.220 192.168.1.220 5100 ctl-trunk stripplus1 Manipto7016 medpool.partnersfb.com 5068 StaticTCP core OPTIONS 30 addplus1 ForEarlyMedia outManipToSFB enabled voip.centurylink.com 257570-4695550016 \*\*\*\*\*\* INVITE addplus1 addplus1 addplus1 stripplus1 stripplus1 stripplus1 core \* 5060 inmanip-before-validate max-udp-length=0 6000 enabled 100rel-interworking ctl-trunk Pass Pass core 192.168.4.135 TCP agents-only enabled 100rel-interworking playback-on-refer="transferrbt" ctl-trunk

o d droo o o	100 160 1 000
	192.100.1.220
allow-anonymous	agents-only
sip-manipulation	
name	Changeinactosendonly
header-rule	
name	changeSDP
header-name	Content-Type
action	manipulate
msg-type	request
methods	INVITE
element-rule	
name	inacttosendonly
parameter-name	application/sdp
type	mime
action	find-replace-all
comparison-type	pattern-rule
match-value	a=inactive
new-value	a=sendonly
sip-manipulation	-
name	Check privacy header
header-rule	encon_privaci_neador
name	ChechForPrivacy
header-name	Privacy
	manipulato
	request
msg-cype	Indust
methods	TNATLE
neader-rule	
name	OverwriteFromDisplay
header-name	From
action	manipulate
comparison-type	boolean
msg-type	request
methods	INVITE
match-value	\$ChechForPrivacy
element-rule	
name	OverwriteUser
type	uri-user
action	find-replace-all
new-value	anonymous
element-rule	
name	OverwriteDisplay
type	uri-display
action	find-replace-all
new-value	"\"Anonymous\" "
sip-manipulation	-
name	ForEarlvMedia
header-rule	-
name	delsupported
header-name	Supported
action	delete
mea-tuno	request
mothoda	TNUTTE
hoador-rula	
neader-rute	
name	adarequireininvitte
header-name	Require
action	add

msg-type request methods INVITE new-value 100rel header-rule name Fortransfer header-name From action sip-manip new-value Changeinactosendonly sip-manipulation Manipto7016 name header-rule modURI name header-name request-uri action manipulate element-rule mod2 name type uri-host action replace match-val-type fqdn \$REMOTE IP+:+\$REMOTE PORT new-value header-rule name DeletePAI header-name P-Asserted-Identity action delete methods INVITE header-rule ReplacePAI name header-name P-Asserted-Identity action add comparison-type pattern-rule methods INVITE new-value sip:4695550016@voip.centurylink.com header-rule name from\_voip\_ctl\_com header-name From action manipulate INVITE methods element-rule From add voip ctl com name type uri-host action replace voip.centurylink.com new-value header-rule registercontactlocalipport name header-name Contact action manipulate request msg-type REGISTER methods element-rule name registercontactlocalip type uri-host action replace match-val-type ip new-value \$LOCAL\_IP element-rule

registercontactlocalport name type uri-port action replace new-value \$LOCAL PORT header-rule name fixContact2 header-name Contact action manipulate element-rule updatecon2 name uri-user type action replace comparison-type pattern-rule match-value (.\*) \$ORIGINAL-^"+1" new-value header-rule fixRoute2 name header-name Route action manipulate element-rule name updateroute2 uri-user type action replace pattern-rule comparison-type match-value (.\*) new-value \$ORIGINAL-^"+1" header-rule name Privacy header-name From action sip-manip new-value Check privacy header sip-manipulation name outManipToSFB header-rule name From header-name From action manipulate msg-type request element-rule name From header type uri-host replace action new-value \$LOCAL IP element-rule modanonfrom name uri-user type action replace comparison-type pattern-rule match-value \+1anonymous new-value anonymous header-rule name То header-name То action manipulate request msg-type

element-rule name То uri-host type action replace new-value \$REMOTE IP header-rule name fixcontactuser header-name Contact action manipulate comparison-type pattern-rule element-rule name user uri-user type action replace comparison-type pattern-rule match-value (.\*) new-value "+1"+\$ORIGINAL sip-monitoring match-any-filter enabled steering-pool ip-address 192.168.1.220 20000 start-port 40000 end-port ctl-trunk realm-id steering-pool ip-address 192.168.4.135 20000 start-port 40000 end-port realm-id core surrogate-agent register-host voip.centurylink.com register-user 4695550016 realm-id core customer-host 192.168.1.220 192.168.1.220 customer-next-hop register-contact-host 192.168.1.220 register-contact-user 4695550016 \*\*\*\*\* password register-expires 60 auth-user 257570-4695550016 system-config hostname CenturyLink-IOT process-log-level DEBUG comm-monitor enabled state monitor-collector 172.18.255.71 address 192.168.1.1 default-gateway translation-rules id addplus1 type add add-string +1 translation-rules id stripplus1 delete type delete-string +1

#### SBC configuration for Single Trunk registration using G729 towards trunk

For this configuration, the SBC will need to transcode between G729 and PCMU codecs on the turnk and SFB side respectively. For transcoding, we configure the codec polices, to-trunk and to-sfb, mentioned below and apply them on the trunk and SFB realms respectively. The following configuration shows the configuration elements that need to be changed for the transcoding configuration to take effect

codec-policy	
name	to-trunk
allow-codecs	* PCMU:no G729::vadoff
add-codecs-on-egress	G729
order-codecs	
packetization-time	20
force-ptime	disabled
dtmf-in-audio	disabled
last-modified-by	admin@172.18.0.139
last-modified-date	2016-08-08 16:43:07
codec-policy	
name	to-sfb
allow-codecs	* G729:no
add-codecs-on-egress	PCMU
order-codecs	
packetization-time	20
force-ptime	disabled
dtmf-in-audio	disabled
last-modified-by	admin@172.18.0.139
last-modified-date	2016-08-08 16:43:07
media-profile	
name	G729
subname	vadoff
payload-type	18
parameters	annexb=no
playback-config	
name	transferg729
entry	
encoding	G729
filename	ringback-us-g729.raw
realm-config	
identifier	core
network-interfaces	s1p0:0
mm-in-realm	enabled
qos-enable	enabled
restricted-latching	sdp
codec-policy	core
realm-config	
identifier	ctl-trunk
network-interfaces	s0p0:0
mm-in-realm	enabled
qos-enable	enabled
media-policy	voip-default
codec-policy	access

# Test Plan Executed

Following is the test plan executed against this setup and results have been documented below.

Title	Description	Test Setup	Status (Passed or Failed etc)
Anonymous Call Rejection Activate	PBX User dials *77 PSTN Calls PBX User with Caller ID Block Should receive an announcement	*77 is Dialed PBX and leaves PBX Phones gets an announcement Calling Party blocks caller ID Calling party makes a call to PBX User Calling Party receives an announcement when PBX user is dialed	Passed
Anonymous Call Rejection Deactivate	PBX User dials *87 PSTN Calls PBX User with Caller ID block Call Should Complete	*87 is dialed PBX User receives and announcement PSTN calls PBX User PSTN Phone receives ringback PBX Phone gets ringing PBX Phone get Caller ID PBX Phone answer the Call 2 way audio is received PBX Phone releases Calls PSTN receives a Bye	Passed
Anonymous Call PBX-BW	PBX sends anonymous call to BW BW delivers the calls Private or unknown or anonymous to PSTN	PBX is configured to send a call to BW as anonymous with TN as PSTN BW delivers the call to PSTN as Private or Anonymous PSTN phone shows the call as Private or Anonymous Call is answered by PSTN PBX user hangs up the call	Passed
Alien TNs	A call PBX call originate where the from TN that is not part of the customer trunk group. As long as the pilot number is identified in outgoing call by PAI, the BroadWorks will accept and route the call.	After Alien TN is set up on a Trunk in CenturyLink Network PBX User Places a Call to PSTN PBX User receives ringback PSTN receives ringing PSTN receives caller id of the Alien TN PSTN answers the call 2 way audio is received PBX Phone releases Calls PSTN receives a Bye	Passed
Barge In	Create a Pick Up Group with 2 PBX Users PSTN Calls PBX User 1 PBX User 2 dials *33 +PBX User Ext PSTN, User 1, and User 2 should be conf	PSTN calls PBX User 1 PSTN Phone receives ringback PBX Phone gets ringing PBX Phone get Caller ID PBX Phone answer the Call 2 way audio is received PBX User 2 Dials *33 + PBX User 1 Extension PSTN, PBX User 1, and PBX User 2 are conferenced together 2 Way Audio is heard by all Legs PBX User 1 drops from Call 2 way Audio is heard by PSTN and PBX User 2 PSTN drops call PBX User 2 receives a Bye	Passed
--------------------	--	--	--------
Barge In Exempt	In the Portal Enable Barge In Exempt Create a Pick Up Group with 2 PBX Users PSTN Calls PBX User 1 PBX User 2 dials *33 +PBX User Ext User 2 Should not be conf	Barge in Exempt is set on PBX user 1 PSTN calls PBX User 1 PSTN Phone receives ringback PBX Phone gets ringing PBX Phone get Caller ID PBX Phone answer the Call 2 way audio is received PBX User 2 Dials *33 + PBX User 1 Extension PBX user 2 is not allowed to barge in PSTN drops the call PBX User 1 receives a Bye	Passed
PSTN to BWA	PSTN calls BWA Number Enter Calling Number (2nd Phone Location) Enter Called Number (PSTN) PSTN should Ring with Caller ID of 2nd Phone Answer Call	BroadWorks Anywhere is set up in Portal PSTN 1 Calls BWA Number Announcement is received Enter calling Number (2nd Phone created in BWA) Announcement received Enter Called Number (PSTN 2) PSTN 1 receives ringback PSTN 2 receives ringing PSTN 2 receives caller ID of 2nd Phone (Not of PSTN 1) PSTN 2 Answers Call 2 way audio is received PSTN 2 releases Calls PSTN receives a Bye	Passed

PSTN to PBX user with BWA	PSTN Calls User with BWA PBX User and 2nd Location should Ring Answer phone for 2nd location	BroadWorks Anywhere is set up in Portal PSTN 1 Calls BWA Number PSTN 1 receives ringback Both PBX User and 2nd Phone Location Number gets ringing Both PBX User and 2nd Phone Location Number gets Caller ID of PSTN Call is answered on Location 2 PBX User no longer gets ringing (cancel) 2 way Audio Location 2 releases call PSTN receives a Bye	Passed
Call Forwarding Always Activate	PBX User dials *72 Enter the CFA Destination TN PSTN calls PBX User with CFA	PBX User 1 Dials *72 Announcement is heard PBX User enter PBX User 2 TN Announcement is heard PBX Receives a Bye PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 does not ring PBX User 2 gets ringing PBX user 2 gets ringing PBX user 2 receives Caller ID (PSTN Originator Caller) PBX User answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed
Call Forwarding Always Interrogate	PBX User with CFA dials *21* Announcement received	PBX User 1 Dials *21* Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed
Call Forwarding Always Deactivate	PBX User with CFA dials *73 PSTN Calls PBX User	PBX User 1 Dials *73 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed

		I	1
PSTN call is CFB to PSTN with ID Restricted	PBX configured to send CFB to BW for identified Station. BW is configured with CFB to PSTN2. PSTN 1 Calls PBX with Caller ID Restricted PSTN 1 hears ring back PBX send 486 Busy to BW BW forwards the call to PSTN2 PSTN 2 hears ringing PSTN 2 Caller ID displays Private/Anonymous PSTN 2 Answers the call. Two way voice path is established between PSTN 1 and PSTN 2 PSTN 2 hangs up	PSTN2 should receive Private/Anonymous as CLID	Passed
PSTN with Privacy call to PBX is CFA to PSTN	PBX User is configured with CFA to PSTN 2 PSTN 1 Calls PBX with Caller ID Restricted PSTN 1 hears ring back PBX sends a new call to BW with PSTN 2 Number, From as Anonymous and PAI set to Pilot Number BW forwards the call to PSTN2 PSTN 2 hears ringing PSTN 2 Caller ID displays Pilot Number PSTN 2 Answers the call. Two way voice path is established between PSTN 1 and PSTN 2 PSTN 2 hangs up	Pilot Number should be shown as CLID on PSTN2	Passed

Call Forwarding Busy Activate	PBX User dials *90 Enter the CFB Destination TN PSTN calls PBX User with CFB	PBX User 1 Dials *90 Announcement is heard PBX User enter PBX User 2 TN Announcement is heard PBX Receives a Bye Busy PBX User 1 PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 does not ring PBX User 2 gets ringing PBX User 2 receives Caller ID (PSTN Originator Caller) PBX User answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed
Call Forwarding Busy Interrogate	PBX User with CFB dials *67* Announcement received	PBX User 1 Dials *67* Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed
Call Forwarding Busy Deactivate	PBX User with CFB dials *91 PSTN Calls PBX User	PBX User 1 Dials *91 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed
Call Forwarding No Answer Activate	PBX User dials *92 Enter the CFNA Destination TN PSTN calls PBX User with CFNA	PBX User 1 Dials *92 Announcement is heard PBX User enters PBX User 2 TN Announcement is heard PBX Receives a Bye PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX User 1 gets ringing PBX User 1 receives Caller ID After timer is RNA is received PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 2 answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed

Call Forwarding No Answer- RNA Timer	PBX User dials *610 Enter 1 # PSTN calls PBX User with CFNA Verify Call is forwarded	PBX User 1 Dials *610 Announcement is Heard PBX User enter 1 for amount of Rings After announcement completes PBX User 1 receives a Bye PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX User 1 gets ringing PBX User 1 receives Caller ID After timer is RNA is received PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 2 answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed
Call Forwarding No Answer Interrogate	PBX User with CFNA dials *61* Announcement received	PBX User 1 Dials *61* Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed
Call Forwarding No Answer Deactivate	PBX User with CFNA dials *93 PSTN Calls PBX User	PBX User 1 Dials *93 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed
Call Forwarding No Answer to Voicemail Activate	PBX User Dials *41 PSTN Dials PBX User with CFNA Verify Call goes to Voicemail	PBX User 1 Dials *41 Announcement is received When announcement completes PBX User receives a Bye Busy PBX User 1 PSTN User Calls PBX User 1 Call should go to voicemail after RNA timer is reached Announcement is Heard Leave voicemail After leaving voicemail PSTN should receive a Bye PBX User 1 should receive and MWI	Passed

Call Forwarding Not Reachable Activate	PBX User dials *94 Enter the CFNR Destination TN Unregister Pilot TNs PSTN calls PBX User with CFNR Verify Call is forwarded Register Pilot TNs	PBX User 1 Dials *94 Announcement is heard PBX User enter PBX User 2 TN Announcement is heard PBX Receives a Bye Unplug SBC Lan Cable PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 does not ring PSTN User 2 gets ringing PSTN User 2 gets ringing PSTN User 2 receives Caller ID (PSTN Originator Caller) PSTN User answers call 2 way Audio PSTN User 1 releases call PSTN User 2 receives a Bye	Passed
Call Forwarding Not Reachable Interrogate	PBX User with CFNR dials *63* Announcement received	PBX User 1 Dials *63* Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed
Call Forwarding Not Reachable Deactivate	PBX User with CFNR dials *95 PSTN Calls PBX User	PBX User 1 Dials *95 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed
Call Return by PBX User	PBX User dials *69	PSTN 1 Calls PBX User 1 PSTN 1 receives ringback PBX User 1 receives caller ID PBX User 1 answers call 2 way Audio PSTN 1 ends the call PBX User 1 receives a Bye PBX User 1 Dials *69 PBX User receives Ringback PSTN 1 receives Ringing PSTN receives Caller ID PSTN answers 2 way Audio PSTN releases call PBX User 1 receives a Bye	Passed

Consultative Transfer with SIP REFER	PBX User Calls PSTN PBX User transfers PSTN to PSTN2 PBX User has Audio with PSTNs PSTN 1 has MOH PBX User Transfers Call PSTN and PSTN2 now have audio		Passed
Unattended Transfer with SIP REFER	PBX User Calls PSTN PBX User transfers PSTN to PSTN2 During Ringback PBX User transfers PSTN 1 has MOH PSTN2 answers call PSTN and PSTN2 now have audio		Passed
Consultative Transfer	PBX User Calls PSTN PBX User transfers PSTN to PBX User 2 PBX User 1 has Audio with PBX User 2 PSTN 1 has MOH PBX User Transfers Call PSTN and PBX 2 now have audio	PBX User 1 Calls PSTN PBX User receives Ringback PSTN 1 receives Ringing PSTN 1 receives Caller ID PSTN 1 answers 2 way Audio PBX User transfers call to PBX User 2 PSTN User gets MOH PBX User 1 gets Dial tone PBX User 1 dials PBX User 2 Extension PBX User 1 receives Ringback PBX User 2 receives Ringing PBX User 2 receives Caller ID of PBX User 1 PBX User 2 answers the Call 2 way Audio PBX User 1 transfers the call MOH Ends PSTN 1 and PBX User 2 are now connected 2 Way Audio PSTN 1 Ends the call PBX User 2 receives the Bye	Passed

Unattended Transfer	PBX User Calls PSTN PBX User transfers PSTN to PBX User 2 During Ringback PBX User transfers PSTN 1 has MOH PBX User 2 answers call PSTN and PBX User 2 now have audio	PBX User 1 Calls PSTN PBX User receives Ringback PSTN 1 receives Ringing PSTN 1 receives Caller ID PSTN 1 answers 2 way Audio PBX User transfers call to PBX User 2 PSTN User gets MOH PBX User 1 gets Dial tone PBX User 1 dials PBX User 2 Extension PBX User 1 receives Ringback PBX User 2 receives Ringback PBX User 2 receives Caller ID of PSTN 1 PBX User 1 release call PBX User 2 answers the Call MOH Ends 2 way Audio PSTN 1 release the call PBX User 2 receives the Bye	Passed
Call Waiting Persistent Activate	PBX User dials *43 PSTN Calls PBX User PSTN 2 Calls PBX User Verify Call Waiting Tone	PBX User 1 Dials *43 Announcement is heard PBX Receives a Bye PSTN User 1 Calls PBX User 1 PSTN User 1 receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN User 2 Calls PBX User 1 PSTN User 1 receives caller ID PBX User 1 hear Call Waiting Tone PBX User 1 hears MOH PBX User 1 hears MOH PBX User 1 answers Call from PSTN 2 2 way Audio Verify PBX User 1 can swap between to callers While on PBX User 1 and PSTN User 1 PSTN 1 releases Call PBX User 1 receives a Bye Call 2 should still be up with PSTN 2 hearing MOH	Passed

Customer Originated Trace	PSTN Calls PBX User PBX User Answers the Call PBX User Hangs up call PBX User enters *57 Verify announcement	PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye PBX User 1 Dial *57 Announcement received Announcement Completes PBX User receives a Bye	Passed
Enhanced Call Logs	Log into portal and verify Call logs	Log into the portal for PBX User 1 On main screen verify calls Logs are displayed Missed Received Placed	Passed
Last Number Redial	PBX User dials *66 The last number dialed should be called	PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases call PBX User 1 receives a Bye PBX User 1 Dial *66 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases call PBX User 1 receives a Bye	Passed

МОН	Verify MOH for conference, transfer, and hold	PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PBX User 1 Places call on Hold PSTN receives MOH PBX User retrieves call from Hold 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed
Remote Office - Like CFA	Provision Remote office for a SIP Trunk user on the BroadWorks portal to use PSTN number A. Place a call from a PSTN number B to the SIP Trunk user's DID and verify that it is forwarded to PSTN number A (the destination configured in BroadWorks).	Log into the portal for PBX User 1 Set up remote Office to PSTN Number 1 PSTN User 2 Calls PBX User 1 PSTN 2 receives ringback PSTN User 1 gets ringing with PSTN 2 Caller ID and Diversion header for PBX User1 PSTN User 1 answers call 2 way Audio PSTN 1 releases call	Passed
Remote Office - Quick Call	Provision Remote office for a SIP Trunk user 1 on the BroadWorks portal to use PSTN number A. On the BW portal, Manage Users, select Configure Features of User 1, under Quick Call, add PSTN B number and click on the Call Button. PSTN A should Start Ringing with PBX User 1 Caller ID.	Log into the portal for PBX User 1 Set up remote Office to PSTN Number 1 Initiate a Quack Call to PSTN 2 on the portal PSTN User 1 gets ringing with PBX User 1 Caller ID PSTN user 1 answers the call. Now PSTN2 should start ringing with PBX User1 as Caller ID. PSTN 1 might hear ringback based on how long PSTN 2 rings. PSTN 2 answers the call 2 way Audio PSTN 1 releases call	Passed

Remote Office - Click to Call	Provision Remote office for a SIP Trunk user 1 on the BroadWorks portal to use PSTN number A. On the BW portal, Manage Users, select Configure Features of User 1, under Call Logs, select either incoming/outgoing/missed calls and Click on a Call under "Phone Number Click To call" column. PSTN A should Start Ringing with PBX User 1 Caller ID.	Log into the portal for PBX User 1 Set up remote Office to PSTN Number 1 Review call logs and identify a call log that needs to be called via Click to Call. Click on the identified call log under "Click to Call" PSTN User 1 gets ringing with PBX User 1 Caller ID PSTN user 1 answers the call. Now PSTN2 should start ringing with PBX User1 as Caller ID. PSTN 1 might hear ringback based on how long PSTN 2 rings. PSTN 2 answers the call 2 way Audio PSTN 1 releases call	Passed
Selective Call Acceptance	Provision selective call acceptance in the BroadWorks portal. Place a call from an accepted TN to the SIP Trunk User. Verify that the call completes normally. Place a call from a TN that is not on the accept list and verify that the call does not reach the SBC.	Log into the portal for PBX User 1 Set up Selected Call Acceptance to PSTN Number 1 PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases Call PBX User 1 receives a Bye	Passed
Selective Call Rejection	Provision selective call rejection in the BroadWorks portal. Place a call from a TN not on the reject list to the SIP Trunk User. Verify that the call completes normally. Place a call from a TN that is on the reject list and verify that the call does not reach the SBC.	Log into the portal for PBX User 1 Set up Selected Call rejection to PSTN Number 1 PSTN Calls PBX User 1 Verify PSTN gets an announcement PSTN receives a Bye	Passed

Sequential Ring	Provision sequential ring in the BroadWorks portal. Place a call to the SIP trunk user. Verify that the numbers in the sequential ring list are dialed in order.	Log into the Portal for PBX User 1 Set up Sequential Ring with PBX User 2 and PBX User 3 PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID After RNA Timer PBX User 1 receives a Cancel PBX User 2 gets ringing PBX user 2 receives Caller ID After RNA Timer PBX User 1 receives a Cancel PBX User 3 gets ringing PBX user 3 receives Caller ID PBX User 3 answers call 2 way Audio PSTN releases Call PBX User 3 receives a Bye	Passed
Simultaneous Ring	Provision Simultaneous ring in the BroadWorks portal. Place a call to the SIP trunk user. Verify that the numbers in the Simultaneous ring list are dialed at once.	Log into the Portal for PBX User 1 Set up Simultaneous Ring with PBX User 2 and PBX User 3 PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 3 gets ringing PBX user 3 receives Caller ID PBX User 3 Answers Call PBX User 1 and 2 receive a Cancel 2 way Audio PSTN releases Call PSTN User 3 receives a Bye	Passed

	I		
PBX Initiate Conference	PBX User Calls PSTN PBX User Conferences PBX User 2	PBX User 1 Calls PSTN PBX User receives Ringback PSTN 1 receives Ringing PSTN 1 receives Caller ID PSTN 1 answers 2 way Audio PBX User conferences call to PBX User 2 PSTN User gets MOH PBX User 1 gets Dial tone PBX User 1 dials PBX User 2 Extension PBX User 1 receives Ringback PBX User 2 receives Ringing PBX User 2 receives Caller ID of PBX User 1 PBX User 2 answers the Call 2 way Audio PBX User 1 conferences the call MOH Ends PSTN 1, PBX User 1 and PBX User 2 are now connected 2 Way Audio PBX User 1 Ends the call PBX User 2 and PSTN receives the Bye	Passed

PSTN Initiate Conference	PBX User calls PSTN PSTN conferences PBX User2	"PBX User 1 Calls PSTN PBX User receives Ringback PSTN 1 receives Ringing PSTN 1 receives Caller ID PSTN 1 answers 2 way Audio PSTN User 1 conferences call to PBX User 2 PBX User 1 gets MOH PSTN User 1 gets Dial tone PSTN User 1 gets Dial tone PSTN User 1 dials PBX User 2 Extension PSTN User 1 receives Ringback PBX User 2 receives Ringback PBX User 2 receives Ringing PBX User 2 receives Caller ID of PSTN User 1 PBX User 2 answers the Call 2 way Audio PSTN User 1 conferences the call MOH Ends PSTN 1, PBX User 1 and PBX User 2 are now connected 2 Way Audio PSTN User 1 Ends the call PBX User 1 and PBX User 2 Still Have Audio PBX User 1 End the Call PBX User 1 End the Call PBX User 1 End the Call PBX User 2 receives a Bye	Passed
Huntgroup Sim Ring	PSTN calls Huntgroup Sim ring 3 members Answer Call	Log into Admin Portal Create Huntgroup with 3 members with Sequential Ring PSTN Calls Huntgroup PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User 2 gets ringing PBX user 2 receives Caller ID PBX user 3 receives Caller ID PBX User 3 Answers the call PBX User 3 Answers the Call 2 way Audio PSTN ends the call PBX User 2 receives a Bye	Pass

PBX to PBX	PBX User Calls PBX User2 Same Trunk Verify RTP is dropped to SBC	PBX User 1 Calls PBX User 2 PBX User 1 receives ringback PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 2 answers call 2 way Audio RTP is on SBC/PBX PBX User 1 End the call PBX User 2 receives a Bye	Passed
PSTN to PBX	PSTN to PBX User	PSTN User 1 Calls PBX User 1 PSTN User 1 receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN User Ends The Call PBX User 1 receives a Bye	Passed
PBX to PSTN	PBX User to PSTN	PBX User 1 Calls PSTN User 1 PBX User 1 receives ringback PSTN User 1 gets ringing PSTN user 1 receives Caller ID PSTN User answers call 2 way Audio PSTN User Ends The Call PBX User 1 receives a Bye	Passed
PBX to PSTN - Packet Marking for SIG packets	PBX to PSTN Call to verify that signaling packets are marked with DSCP = 24 (0x18)	All outgoing SIP Signaling packets are marked with DSCP=24	Passed
PBX to PSTN - Packet Marking for RTP packets	PBX to PSTN Call to verify that rtp packets are marked with DSCP = 40 (0x28)	All outgoing SIP RTP packets are marked with DSCP=40	Passed
PBX to PSTN - Directory assistance	PBX User Calls PBX 411 and speaks with directory assistant	PBX User 1 dials 411 Call is delivered to Directory Assistant for enquiry Once the user hears an announcement or speaks with an operator, PBX user hangs up the call	Passed
PBX to PSTN - Toll Free	PBX User Calls 800.366.8201 to test toll free numbers	PBX User 1 dials 800.366.8201 (CTL Support) Call is delivered to CenturyLink Support Once the user hears an announcement or speaks with an operator, PBX user hangs up the call	Passed

Anonymous Call Rejection Activate	PBX User dials *77 PSTN Calls PBX User with Caller ID Block Should receive an announcement	*77 is Dialed PBX and leaves PBX Phones gets an announcement Calling Party blocks caller ID Calling party makes a call to PBX User Calling Party receives an announcement when PBX user is dialed	Passed
Anonymous Call Rejection Deactivate	PBX User dials *87 PSTN Calls PBX User with Caller ID block Call Should Complete	*87 is dialed PBX User receives and announcement PSTN calls PBX User PSTN Phone receives ringback PBX Phone gets ringing PBX Phone get Caller ID PBX Phone answer the Call 2 way audio is received PBX Phone releases Calls PSTN receives a Bye	Passed
Anonymous Call PBX-BW	PBX sends anonymous call to BW BW delivers the calls Private or unknown or anonymous to PSTN	PBX is configured to send a call to BW as anonymous with TN as PSTN BW delivers the call to PSTN as Private or Anonymous PSTN phone shows the call as Private or Anonymous Call is answered by PSTN PBX user hangs up the call	Passed
Alien TNs	A call PBX call originate where the from TN that is not part of the customer trunk group. As long as the pilot number is identified in outgoing call by PAI, the BroadWorks will accept and route the call.	After Alien TN is set up on a Trunk in CenturyLink Network PBX User Places a Call to PSTN PBX User receives ringback PSTN receives ringing PSTN receives caller id of the Alien TN PSTN answers the call 2 way audio is received PBX Phone releases Calls PSTN receives a Bye	Passed

Barge In	Create a Pick Up Group with 2 PBX Users PSTN Calls PBX User 1 PBX User 2 dials *33 +PBX User Ext PSTN, User 1, and User 2 should be conf	PSTN calls PBX User 1 PSTN Phone receives ringback PBX Phone gets ringing PBX Phone get Caller ID PBX Phone answer the Call 2 way audio is received PBX User 2 Dials *33 + PBX User 1 Extension PSTN, PBX User 1, and PBX User 2 are conferenced together 2 Way Audio is heard by all Legs PBX User 1 drops from Call 2 way Audio is heard by PSTN and PBX User 2	Passed
		PSTN drops call PBX User 2 receives a Bye	
Barge In Exempt	In the Portal Enable Barge In Exempt Create a Pick Up Group with 2 PBX Users PSTN Calls PBX User 1 PBX User 2 dials *33 +PBX User Ext User 2 Should not be conf	Barge in Exempt is set on PBX user 1 PSTN calls PBX User 1 PSTN Phone receives ringback PBX Phone gets ringing PBX Phone get Caller ID PBX Phone answer the Call 2 way audio is received PBX User 2 Dials *33 + PBX User 1 Extension PBX user 2 is not allowed to barge in PSTN drops the call PBX User 1 receives a Bye	Passed
PSTN to BWA	PSTN calls BWA Number Enter Calling Number (2nd Phone Location) Enter Called Number (PSTN) PSTN should Ring with Caller ID of 2nd Phone Answer Call	BroadWorks Anywhere is set up in Portal PSTN 1 Calls BWA Number Announcement is received Enter calling Number (2nd Phone created in BWA) Announcement received Enter Called Number (PSTN 2) PSTN 1 receives ringback PSTN 2 receives ringing PSTN 2 receives caller ID of 2nd Phone (Not of PSTN 1) PSTN 2 Answers Call 2 way audio is received PSTN 2 releases Calls PSTN receives a Bye	Passed

PSTN to PBX user with BWA	PSTN Calls User with BWA PBX User and 2nd Location should Ring Answer phone for 2nd location	BroadWorks Anywhere is set up in Portal PSTN 1 Calls BWA Number PSTN 1 receives ringback Both PBX User and 2nd Phone Location Number gets ringing Both PBX User and 2nd Phone Location Number gets Caller ID of PSTN Call is answered on Location 2 PBX User no longer gets ringing (cancel) 2 way Audio Location 2 releases call PSTN receives a Bye	Passed
Call Forwarding Always Activate	PBX User dials *72 Enter the CFA Destination TN PSTN calls PBX User with CFA	PBX User 1 Dials *72 Announcement is heard PBX User enter PBX User 2 TN Announcement is heard PBX Receives a Bye PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 does not ring PBX User 2 gets ringing PBX user 2 gets ringing PBX user 2 receives Caller ID (PSTN Originator Caller) PBX User answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed
Call Forwarding Always Interrogate	PBX User with CFA dials *21* Announcement received	PBX User 1 Dials *21* Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed
Call Forwarding Always Deactivate	PBX User with CFA dials *73 PSTN Calls PBX User	PBX User 1 Dials *73 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed

PSTN with Privacy call to PBX is CFA to PSTN	PBX User is configured with CFA to PSTN 2 PSTN 1 Calls PBX with Caller ID Restricted PSTN 1 hears ring back PBX sends a new call to BW with PSTN 2 Number, From as Anonymous and PAI set to Pilot Number BW forwards the call to PSTN2 PSTN 2 hears ringing PSTN 2 Caller ID displays Pilot Number PSTN 2 Answers the call. Two way voice path is established between PSTN 1 and PSTN 2 PSTN 2 hangs up	Pilot Number should be shown as CLID on PSTN2	Passed
PSTN call is CFB to PSTN with ID Restricted	PBX configured to send CFB to BW for identified Station. BW is configured with CFB to PSTN2. PSTN 1 Calls PBX with Caller ID Restricted PSTN 1 hears ring back PBX send 486 Busy to BW BW forwards the call to PSTN2 PSTN 2 hears ringing PSTN 2 Caller ID displays Private/Anonymous PSTN 2 Answers the call. Two way voice path is established between PSTN 1 and PSTN 2 PSTN 2 hangs up	PSTN2 should receive Private/Anonymous as CLID	Passed

Call Forwarding Busy Activate	PBX User dials *90 Enter the CFB Destination TN PSTN calls PBX User with CFB	PBX User 1 Dials *90 Announcement is heard PBX User enter PBX User 2 TN Announcement is heard PBX Receives a Bye Busy PBX User 1 PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 does not ring PBX User 2 gets ringing PBX User 2 gets ringing PBX user 2 receives Caller ID (PSTN Originator Caller) PBX User answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed
Call Forwarding Busy Interrogate	PBX User with CFB dials *67* Announcement received	PBX User 1 Dials *67* Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed
Call Forwarding Busy Deactivate	PBX User with CFB dials *91 PSTN Calls PBX User	PBX User 1 Dials *91 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed
Call Forwarding No Answer Activate	PBX User dials *92 Enter the CFNA Destination TN PSTN calls PBX User with CFNA	PBX User 1 Dials *92 Announcement is heard PBX User enters PBX User 2 TN Announcement is heard PBX Receives a Bye PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX User 1 gets ringing PBX User 1 receives Caller ID After timer is RNA is received PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 2 answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed

Call Forwarding No Answer- RNA Timer	PBX User dials *610 Enter 1 # PSTN calls PBX User with CFNA Verify Call is forwarded	PBX User 1 Dials *610 Announcement is Heard PBX User enter 1 for amount of Rings After announcement completes PBX User 1 receives a Bye PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX User 1 receives Caller ID After timer is RNA is received PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 2 answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed
Call Forwarding No Answer Interrogate	PBX User with CFNA dials *61* Announcement received	PBX User 1 Dials *61* Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed
Call Forwarding No Answer Deactivate	PBX User with CFNA dials *93 PSTN Calls PBX User	PBX User 1 Dials *93 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed
Call Forwarding No Answer to Voicemail Activate	PBX User Dials *41 PSTN Dials PBX User with CFNA Verify Call goes to Voicemail	PBX User 1 Dials *41 Announcement is received When announcement completes PBX User receives a Bye Busy PBX User 1 PSTN User Calls PBX User 1 Call should go to voicemail after RNA timer is reached Announcement is Heard Leave voicemail After leaving voicemail PSTN should receive a Bye PBX User 1 should receive and MWI	Passed

Call Forwarding Not Reachable Activate	PBX User dials *94 Enter the CFNR Destination TN Unregister Pilot TNs PSTN calls PBX User with CFNR Verify Call is forwarded Register Pilot TNs	PBX User 1 Dials *94 Announcement is heard PBX User enter PBX User 2 TN Announcement is heard PBX Receives a Bye Unplug SBC Lan Cable PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 does not ring PSTN User 2 gets ringing PSTN User 2 gets ringing PSTN user 2 receives Caller ID (PSTN Originator Caller) PSTN User answers call 2 way Audio PSTN User 1 releases call PSTN User 2 receives a Bye	Passed
Call Forwarding Not Reachable Interrogate	PBX User with CFNR dials *63* Announcement received	PBX User 1 Dials *63* Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed
Call Forwarding Not Reachable Deactivate	PBX User with CFNR dials *95 PSTN Calls PBX User	PBX User 1 Dials *95 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed
Call Return by PBX User	PBX User dials *69	PSTN 1 Calls PBX User 1 PSTN 1 receives ringback PBX User 1 receives caller ID PBX User 1 answers call 2 way Audio PSTN 1 ends the call PBX User 1 receives a Bye PBX User 1 Dials *69 PBX User receives Ringback PSTN 1 receives Ringing PSTN receives Caller ID PSTN answers 2 way Audio PSTN releases call PBX User 1 receives a Bye	Passed

Consultative Transfer with SIP REFER	PBX User Calls PSTN PBX User transfers PSTN to PSTN2 PBX User has Audio with PSTNs PSTN 1 has MOH PBX User Transfers Call PSTN and PSTN2 now have audio		Passed
Unattended Transfer with SIP REFER	PBX User Calls PSTN PBX User transfers PSTN to PSTN2 During Ringback PBX User transfers PSTN 1 has MOH PSTN2 answers call PSTN and PSTN2 now have audio		Passed
Consultative Transfer	PBX User Calls PSTN PBX User transfers PSTN to PBX User 2 PBX User 1 has Audio with PBX User 2 PSTN 1 has MOH PBX User Transfers Call PSTN and PBX 2 now have audio	PBX User 1 Calls PSTN PBX User receives Ringback PSTN 1 receives Ringing PSTN 1 receives Caller ID PSTN 1 answers 2 way Audio PBX User transfers call to PBX User 2 PSTN User gets MOH PBX User 1 gets Dial tone PBX User 1 dials PBX User 2 Extension PBX User 1 receives Ringback PBX User 2 receives Ringing PBX User 2 receives Caller ID of PBX User 1 PBX User 2 answers the Call 2 way Audio PBX User 1 transfers the call MOH Ends PSTN 1 and PBX User 2 are now connected 2 Way Audio PSTN 1 Ends the call PBX User 2 receives the Bye	Passed

	r		1
Unattended Transfer	PBX User Calls PSTN PBX User transfers PSTN to PBX User 2 During Ringback PBX User transfers PSTN 1 has MOH PBX User 2 answers call PSTN and PBX User 2 now have audio	PBX User 1 Calls PSTN PBX User receives Ringback PSTN 1 receives Ringing PSTN 1 receives Caller ID PSTN 1 answers 2 way Audio PBX User transfers call to PBX User 2 PSTN User gets MOH PBX User 1 gets Dial tone PBX User 1 dials PBX User 2 Extension PBX User 1 receives Ringback PBX User 2 receives Ringing PBX User 2 receives Caller ID of PSTN 1 PBX User 1 release call PBX User 2 answers the Call MOH Ends 2 way Audio PSTN 1 release the call PBX User 2 receives the Bye	Passed
Call Waiting Persistent Activate	PBX User dials *43 PSTN Calls PBX User PSTN 2 Calls PBX User Verify Call Waiting Tone	PBX User 1 Dials *43 Announcement is heard PBX Receives a Bye PSTN User 1 Calls PBX User 1 PSTN User 1 receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN User 2 Calls PBX User 1 PSTN User 1 receives caller ID PBX User 1 hear Call Waiting Tone PBX User 1 hears Call Waiting Tone PBX User 1 hears MOH PBX User 1 answers Call from PSTN 2 2 way Audio Verify PBX User 1 can swap between to callers While on PBX User 1 and PSTN User 1 PSTN 1 releases Call PBX User 1 receives a Bye Call 2 should still be up with PSTN 2 hearing MOH	Passed
Customer Originated Trace	PSTN Calls PBX User PBX User Answers the Call PBX User Hangs up call PBX User enters *57 Verify announcement	PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye PBX User 1 Dial *57 Announcement received Announcement Completes PBX User receives a Bye	Passed

Enhanced Call Logs	Log into portal and verify Call logs	Log into the portal for PBX User 1 On main screen verify calls Logs are displayed Missed Received Placed	Passed
Last Number Redial	PBX User dials *66 The last number dialed should be called	PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases call PBX User 1 receives a Bye PBX User 1 Dial *66 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases call PBX User 1 receives a Bye	Passed
МОН	Verify MOH for conference, transfer, and hold	PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PBX User 1 Places call on Hold PSTN receives MOH PBX User retrieves call from Hold 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed

Remote Office - Like CFA	Provision Remote office for a SIP Trunk user on the BroadWorks portal to use PSTN number A. Place a call from a PSTN number B to the SIP Trunk user's DID and verify that it is forwarded to PSTN number A (the destination configured in BroadWorks).	Log into the portal for PBX User 1 Set up remote Office to PSTN Number 1 PSTN User 2 Calls PBX User 1 PSTN 2 receives ringback PSTN User 1 gets ringing with PSTN 2 Caller ID and Diversion header for PBX User1 PSTN User 1 answers call 2 way Audio PSTN 1 releases call	Passed
Remote Office - Quick Call	Provision Remote office for a SIP Trunk user 1 on the BroadWorks portal to use PSTN number A. On the BW portal, Manage Users, select Configure Features of User 1, under Quick Call, add PSTN B number and click on the Call Button. PSTN A should Start Ringing with PBX User 1 Caller ID.	Log into the portal for PBX User 1 Set up remote Office to PSTN Number 1 Initiate a Quick Call to PSTN 2 on the portal PSTN User 1 gets ringing with PBX User 1 Caller ID PSTN user 1 answers the call. Now PSTN2 should start ringing with PBX User1 as Caller ID. PSTN 1 might hear ringback based on how long PSTN 2 rings. PSTN 2 answers the call 2 way Audio PSTN 1 releases call	Passed
Remote Office - Click to Call	Provision Remote office for a SIP Trunk user 1 on the BroadWorks portal to use PSTN number A. On the BW portal, Manage Users, select Configure Features of User 1, under Call Logs, select either incoming/outgoing/missed calls and Click on a Call under "Phone Number Click To call" column. PSTN A should Start Ringing with PBX User 1 Caller ID.	Log into the portal for PBX User 1 Set up remote Office to PSTN Number 1 Review call logs and identify a call log that needs to be called via Click to Call. Click on the identified call log under "Click to Call" PSTN User 1 gets ringing with PBX User 1 Caller ID PSTN user 1 answers the call. Now PSTN2 should start ringing with PBX User1 as Caller ID. PSTN 1 might hear ringback based on how long PSTN 2 rings. PSTN 2 answers the call 2 way Audio PSTN 1 releases call	Passed

Selective Call Acceptance	Provision selective call acceptance in the BroadWorks portal. Place a call from an accepted TN to the SIP Trunk User. Verify that the call completes normally. Place a call from a TN that is not on the accept list and verify that the call does not reach the SBC.	Log into the portal for PBX User 1 Set up Selected Call Acceptance to PSTN Number 1 PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases Call PBX User 1 receives a Bye	Passed
Selective Call Rejection	Provision selective call rejection in the BroadWorks portal. Place a call from a TN not on the reject list to the SIP Trunk User. Verify that the call completes normally. Place a call from a TN that is on the reject list and verify that the call does not reach the SBC.	Log into the portal for PBX User 1 Set up Selected Call rejection to PSTN Number 1 PSTN Calls PBX User 1 Verify PSTN gets an announcement PSTN receives a Bye	Passed
Sequential Ring	Provision sequential ring in the BroadWorks portal. Place a call to the SIP trunk user. Verify that the numbers in the sequential ring list are dialed in order.	Log into the Portal for PBX User 1 Set up Sequential Ring with PBX User 2 and PBX User 3 PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID After RNA Timer PBX User 1 receives a Cancel PBX User 2 gets ringing PBX user 2 receives Caller ID After RNA Timer PBX User 1 receives a Cancel PBX User 3 gets ringing PBX user 3 receives Caller ID PBX User 3 answers call 2 way Audio PSTN releases Call PBX User 3 receives a Bye	Passed

Simultaneous Ring	Provision Simultaneous ring in the BroadWorks portal. Place a call to the SIP trunk user. Verify that the numbers in the Simultaneous ring list are dialed at once.	Log into the Portal for PBX User 1 Set up Simultaneous Ring with PBX User 2 and PBX User 3 PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 3 gets ringing PBX user 3 receives Caller ID PBX User 3 Answers Call PBX User 1 and 2 receive a Cancel 2 way Audio PSTN releases Call PSTN User 3 receives a Bye	Passed
PBX Initiate Conference	PBX User Calls PSTN PBX User Conferences PBX User 2	PBX User 1 Calls PSTN PBX User receives Ringback PSTN 1 receives Ringing PSTN 1 receives Caller ID PSTN 1 answers 2 way Audio PBX User conferences call to PBX User 2 PSTN User gets MOH PBX User 1 gets Dial tone PBX User 1 dials PBX User 2 Extension PBX User 1 receives Ringback PBX User 2 receives Caller ID of PBX User 1 PBX User 2 answers the Call 2 way Audio PBX User 1 conferences the call MOH Ends PSTN 1, PBX User 1 and PBX User 2 are now connected 2 Way Audio PBX User 1 Ends the call PBX User 2 and PSTN receives the Bye	Passed

PSTN Initiate Conference	PBX User calls PSTN PSTN conferences PBX User2	PBX User 1 Calls PSTN PBX User receives Ringback PSTN 1 receives Ringing PSTN 1 receives Caller ID PSTN 1 answers 2 way Audio PSTN User 1 conferences call to PBX User 2 PBX User 1 gets MOH PSTN User 1 gets Dial tone PSTN User 1 gets Dial tone PSTN User 1 dials PBX User 2 Extension PSTN User 1 receives Ringback PBX User 2 receives Ringing PBX User 2 receives Caller ID of PSTN User 1 PBX User 2 answers the Call 2 way Audio PSTN User 1 conferences the call MOH Ends PSTN 1, PBX User 1 and PBX User 2 are now connected 2 Way Audio PSTN User 1 Ends the call PBX User 1 and PBX User 2 Still Have Audio PBX User 1 End the Call PBX User 2 receives a Bye	Passed
Huntgroup Sim Ring	PSTN calls Huntgroup Sim ring 3 members Answer Call	Log into Admin Portal Create Huntgroup with 3 members with Sequential Ring PSTN Calls Huntgroup PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User 2 gets ringing PBX user 2 receives Caller ID PBX user 3 receives Caller ID PBX User 3 Answers the call PBX User 3 Answers the Call 2 way Audio PSTN ends the call PBX User 2 receives a Bye	Passed
PBX to PBX	PBX User Calls PBX User2 Same Trunk Verify RTP is dropped to SBC	PBX User 1 Calls PBX User 2 PBX User 1 receives ringback PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 2 answers call 2 way Audio RTP is on SBC/PBX PBX User 1 End the call PBX User 2 receives a Bye	Passed

PSTN to PBX	PSTN to PBX User	PSTN User 1 Calls PBX User 1 PSTN User 1 receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN User Ends The Call PBX User 1 receives a Bye	Passed
PBX to PSTN	PBX User to PSTN	PBX User 1 Calls PSTN User 1 PBX User 1 receives ringback PSTN User 1 gets ringing PSTN user 1 receives Caller ID PSTN User answers call 2 way Audio PSTN User Ends The Call PBX User 1 receives a Bye	Passed
PBX to PSTN - Packet Marking for SIG packets	PBX to PSTN Call to verify that signaling packets are marked with DSCP = 24 (0x18)	All outgoing SIP Signaling packets are marked with DSCP=24	Passed
PBX to PSTN - Packet Marking for RTP packets	PBX to PSTN Call to verify that rtp packets are marked with DSCP = 40 (0x28)	All outgoing SIP RTP packets are marked with DSCP=40	Passed
PBX to PSTN - Directory assistance	PBX User Calls PBX 411 and speaks with directory assistant	PBX User 1 dials 411 Call is delivered to Directory Assistant for enquiry Once the user hears an announcement or speaks with an operator, PBX user hangs up the call	Passed
PBX to PSTN - Toll Free	PBX User Calls 800.366.8201 to test toll free numbers	PBX User 1 dials 800.366.8201 (CTL Support) Call is delivered to CenturyLink Support Once the user hears an announcement or speaks with an operator, PBX user hangs up the call	Passed

## Configuration for Dual trunk registration and test plan

In the dual trunk architecture, the ESBC registers to a pair of geo-redundant CTL carriers SBC. The architecture for dual trunk registration is as follows:



This architecture consists of a single SBC with two realms on each side of the SBC. Two realms and two sip-interfaces on each side replicate two trunks on each side. This test plan, like the single trunk registration, was tested for both PCMU and G729.

obo configuration for baar trank registration asing romo towards train	SBC	configuration	for Dual	Trunk reg	gistration	using	PCMU	towards	trun
--	-----	---------------	----------	-----------	------------	-------	------	---------	------

local-policy		
from-address	*	
to-address	*	
source-realm	ctl-trunk2	
policy-attribute		
next-hop	sag:SFBmed	
realm	core2	
local-policy		
from-address	*	
to-address	*	
source-realm	core	
policy-attribute		
next-hop	192.168.1.170	
realm	ctl-trunk	
local-policy		
from-address	*	
to-address	SFBMed1.partnersfb.com	
source-realm	core	
policy-attribute		
next-hop	SFBMed1.partnersfb.com	
realm	core	

action replace-uri local-policy from-address + to-address SFBMed2.partnersfb.com source-realm core policy-attribute next-hop SFBMed2.partnersfb.com realm core action replace-uri local-policy from-address 4695550016 to-address source-realm core policy-attribute 192.168.1.170 next-hop realm ctl-trunk local-policy from-address \* to-address source-realm core2 policy-attribute 192.168.1.171 next-hop realm ctl-trunk2 local-policy from-address to-address SFBMed1.partnersfb.com source-realm core2 policy-attribute SFBMed1.partnersfb.com next-hop realm core2 action replace-uri local-policy from-address to-address SFBMed2.partnersfb.com source-realm core2 policy-attribute SFBMed2.partnersfb.com next-hop realm core2 action replace-uri local-policy from-address 4695737015 to-address source-realm core2 policy-attribute next-hop 192.168.1.171 ctl-trunk2 realm local-policy from-address to-address source-realm ctl-trunk policy-attribute next-hop medpool.partnersfb.com realm core media-manager media-policy voip-default name

too ootting	
tos-settings	audio
niedza-type	auaro 020
tog-gottingg	0X20
media-ture	message
media-sub-type	sin
tos-value	0x18
	UALU
network-interface	
name	s0p0
ip-address	192.168.1.220
netmask	255.255.255.0
gateway	192.168.1.1
hip-ip-list	192.168.1.220
	192.168.1.221
icmp-address	192.168.1.220
	192.168.1.221
network-interface	
name	s1p0
hostname	attsbc.partnersib.com
1p-address	192.168.4.135
netmask	
dag-in-primary	192.100.4.1
dns-domain	192.100.4.130
hin-in-list	192 168 1 135
	192.168.4.136
icmp-address	192.168.4.135
	192.168.4.136
network-interface	
name	s1p1
ip-address	192.168.3.220
netmask	255.255.255.0
gateway	192.168.3.1
hip-ip-list	192.168.3.220
icmp-address	192.168.3.220
phy-interface	
name	50p0
operation-type	Media
phy-interface	
name	slp0
operation-type	Media
slot	T
pny-interface	c1m1
name	SIDI
operation-type	1
slot	1
playback-config	1
name	transferrht
entry	
encodina	PCMU
filename	US ringbackPCMU.raw
realm-config	
identifier	ctl-trunk2

network-interfaces s0p0:0 mm-in-realm enabled enabled gos-enable media-policy voip-default realm-config identifier core network-interfaces s1p0:0 mm-in-realm enabled enabled qos-enable restricted-latching sdp realm-config identifier core2 network-interfaces s1p0:0 mm-in-realm enabled qos-enable enabled realm-config identifier ctl-trunk network-interfaces s0p0:0 mm-in-realm enabled qos-enable enabled media-policy voip-default session-agent 192.168.1.171 hostname 192.168.1.171 ip-address 5100 port realm-id ctl-trunk2 out-translationid stripplus1 out-manipulationid Manipto7015 session-agent hostname 192.168.1.170 192.168.1.170 ip-address 5100 port ctl-trunk realm-id out-translationid stripplus1 out-manipulationid Manipto7016 session-agent SFBMed1.partnersfb.com hostname ip-address 192.168.4.152 5069 port transport-method StaticTCP realm-id core2 ping-method OPTIONS ping-interval 30 out-translationid addplus1 ForEarlyMedia in-manipulationid out-manipulationid outManipToSFB refer-call-transfer enabled auth-attributes auth-realm voip.centurylink.com username 257570-4695737015 password \*\*\*\*\*\* in-dialog-methods INVITE session-agent hostname SFBMed2.partnersfb.com 192.168.4.153 ip-address

5069 port transport-method realm-id core2 ping-method ping-interval 30 out-translationid in-manipulationid out-manipulationid refer-call-transfer auth-attributes auth-realm username password in-dialog-methods session-agent hostname 5068 port transport-method realm-id core ping-method ping-interval 30 out-translationid in-manipulationid out-manipulationid refer-call-transfer auth-attributes auth-realm username password in-dialog-methods session-group group-name SFBmed strategy dest session-group ctl group-name dest session-translation id rules-calling rules-called session-translation id rules-calling rules-called sip-config home-realm-id core registrar-domain \* \* registrar-host registrar-port 5060 options max-udp-length=0 sip-message-len 6000 refer-src-routing enabled

StaticTCP OPTIONS addplus1 ForEarlyMedia outManipToSFB enabled voip.centurylink.com 257570-4695737015 \*\*\*\*\*\* INVITE medpool.partnersfb.com StaticTCP OPTIONS addplus1 ForEarlyMedia outManipToSFB enabled voip.centurylink.com 257570-4695550016 \*\*\*\*\*\* INVITE RoundRobin SFBMed1.partnersfb.com SFBMed2.partnersfb.com 192.168.1.171 192.168.1.170 addplus1 addplus1 addplus1 stripplus1 stripplus1 stripplus1 inmanip-before-validate

sip-feature 100rel-interworking name ctl-trunk realm require-mode-inbound Pass require-mode-outbound Pass sip-interface realm-id ctl-trunk2 sip-port 192.168.1.221 address allow-anonymous agents-only sip-interface realm-id core sip-port 192.168.4.135 address transport-protocol TCP allow-anonymous agents-only registration-caching enabled options 100rel-interworking spl-options playback-on-refer="transferrbt" sip-interface realm-id core2 sip-port 192.168.4.136 address transport-protocol TCP registration-caching enabled 100rel-interworking options spl-options playback-on-refer="transferrbt" sip-interface realm-id ctl-trunk sip-port 192.168.1.220 address allow-anonymous agents-only sip-manipulation Changeinactosendonly name header-rule name changeSDP header-name Content-Type action manipulate request msg-type methods INVITE element-rule name inacttosendonly parameter-name application/sdp mime type find-replace-all action pattern-rule comparison-type match-value a=inactive new-value a=sendonly sip-manipulation name Check privacy header header-rule name ChechForPrivacy header-name Privacy action manipulate request msg-type
methoda.	T \17.7 mm
methods	TUATLE
neader-rule	
name	OverwriteFromDisplay
header-name	From
action	manipulate
comparison-type	boolean
msg-type	request
methods	INVITE
match-value	\$ChechForPrivacy
element-rule	
name	OverwriteUser
type	uri-user
action	find-replace-all
new-value	anonymous
element-rule	
name	OverwriteDisplay
type	uri-display
action	find-replace-all
new-value	"\"Anonymous\" "
sip-manipulation	
name	ForEarlyMedia
header-rule	
name	delsupported
header-name	Supported
action	delete
msg-type	request
methods	TNVTTE
header-rule	
name	addrequireinINVITE
header-name	Require
action	add
msg-tupe	request
most bods	
	100rol
header-rule	TOOLET
neader-rure	Fortranafor
handor-nomo	FOICIANSIEL
	rio
	Change in estagender lu
new-value	changernaccosendonry
sip-manipulation	No
name	Manipto/015
neader-rule	
name	modURI
header-name	request-uri
action	manipulate
element-rule	
name	mod2
type	uri-host
action	replace
match-val-type	fqdn
new-value	<pre>\$REMOTE_IP+:+\$REMOTE_PORT</pre>
header-rule	
name	DeletePAI
header-name	P-Asserted-Identity
action	delete
methods	INVITE

header-rule name ReplacePAI header-name action add comparison-type methods INVITE new-value header-rule name header-name From action manipulate INVITE methods element-rule name type action new-value header-rule name header-name Contact action manipulate msg-type request methods REGISTER element-rule name type action match-val-type ip new-value element-rule name type action new-value header-rule name fixContact1 header-name Contact action manipulate element-rule name type action comparison-type match-value new-value header-rule fixRoute1 name header-name Route action manipulate element-rule name type action comparison-type pattern-rule match-value (.\*)

new-value

P-Asserted-Identity pattern-rule sip:4695737015@voip.centurylink.com from\_voip\_ctl\_com From\_add\_voip\_ctl\_com uri-host replace voip.centurylink.com registercontactlocalipport registercontactlocalip uri-host replace \$LOCAL\_IP registercontactlocalport uri-port replace \$LOCAL PORT updatecon uri-user replace pattern-rule (.\*) \$ORIGINAL-^"+1" updateroute uri-user replace

\$ORIGINAL-^"+1"

header-rule name Privacy header-name From sip-manip action new-value Check\_privacy\_header sip-manipulation name Manipto7016 header-rule name modURI header-name request-uri action manipulate element-rule mod2 name uri-host type action replace match-val-type fqdn \$REMOTE\_IP+:+\$REMOTE\_PORT new-value header-rule name DeletePAI header-name P-Asserted-Identity action delete methods INVITE header-rule ReplacePAI name P-Asserted-Identity header-name action add comparison-type pattern-rule methods INVITE new-value sip:4695550016@voip.centurylink.com header-rule name from voip ctl com header-name From action manipulate methods INVITE element-rule name From\_add\_voip\_ctl\_com uri-host type replace action new-value voip.centurylink.com header-rule name registercontactlocalipport header-name Contact action manipulate msg-type request methods REGISTER element-rule registercontactlocalip name uri-host type action replace match-val-type ip new-value \$LOCAL IP element-rule name registercontactlocalport type uri-port action replace \$LOCAL PORT new-value

header-rule	
name	fixContact2
header-name	Contact
action	manipulate
element-rule	
name	updatecon2
type	uri-user
action	replace
comparison-type	pattern-rule
match-value	(.*)
new-value	\$ORIGINAL-^"+1"
header-rule	
name	fixRoute2
neader-name	Route
action	manipulace
erement-rure	undata noute 2
tuno	updateroutez
cype	
action	reprace
match_value	( *)
	() SORICINAL - ^ "+1 "
header-rule	ÇONTGINAL TI
name	Privacy
header-name	From
action	sip-manip
new-value	Check privacy header
sip-manipulation	
name	outManipToSFB
header-rule	
name	From
header-name	From
action	manipulate
msg-type	request
element-rule	
name	From_header
type	uri-host
action	replace
new-value	\$LOCAL_IP
element-rule	
name	modanonirom
type	uri-user
action	replace
comparison-type	pattern-rule
nateri-value	
header-rule	anonymous
name	To
header-name	To
action	manipulate
msg-type	request
element-rule	2044000
name	То
type	uri-host

new-value header-rule name header-name Contact action comparison-type element-rule name type action comparison-type match-value new-value sip-monitoring enabled match-any-filter steering-pool ip-address 192.168.1.220 20000 start-port 40000 end-port realm-id ctl-trunk steering-pool 192.168.1.221 ip-address start-port 20000 end-port 40000 ctl-trunk2 realm-id steering-pool 192.168.3.220 ip-address 20000 start-port end-port 40000 realm-id Nice-SIPREC steering-pool 192.168.4.135 ip-address 20000 start-port end-port 40000 realm-id core steering-pool ip-address 192.168.4.136 20000 start-port end-port 40000 realm-id core2 surrogate-agent register-host voip.centurylink.com 4695737015 register-user realm-id core2 customer-host 192.168.1.171 192.168.1.171 customer-next-hop register-contact-host 192.168.1.171 register-contact-user 4695737015 \*\*\*\*\* password register-expires 60 auth-user 257570-4695737015 surrogate-agent

register-host

register-user

#### \$REMOTE IP

fixcontactuser manipulate pattern-rule

voip.centurylink.com

4695550016

user uri-user replace pattern-rule (.\*) "+1"+\$ORIGINAL

realm-id	core
customer-host	192.168.1.170
customer-next-hop	192.168.1.170
register-contact-host	192.168.1.170
register-contact-user	4695550016
password	****
register-expires	60
auth-user	257570-4695550016
system-config	
hostname	CenturyLink-IOT
process-log-level	DEBUG
comm-monitor	
state	enabled
state monitor-collector	enabled
state monitor-collector address	enabled 172.18.255.71
state monitor-collector address default-gateway	enabled 172.18.255.71 192.168.1.1
state monitor-collector address default-gateway translation-rules	enabled 172.18.255.71 192.168.1.1
state monitor-collector address default-gateway translation-rules id	enabled 172.18.255.71 192.168.1.1 addplus1
state monitor-collector address default-gateway translation-rules id type	enabled 172.18.255.71 192.168.1.1 addplus1 add
state monitor-collector address default-gateway translation-rules id type add-string	enabled 172.18.255.71 192.168.1.1 addplus1 add +1
state monitor-collector address default-gateway translation-rules id type add-string translation-rules	enabled 172.18.255.71 192.168.1.1 addplus1 add +1
state monitor-collector address default-gateway translation-rules id type add-string translation-rules id	enabled 172.18.255.71 192.168.1.1 addplus1 add +1 stripplus1
state monitor-collector address default-gateway translation-rules id type add-string translation-rules id type	enabled 172.18.255.71 192.168.1.1 addplus1 add +1 stripplus1 delete
state monitor-collector address default-gateway translation-rules id type add-string translation-rules id type delete-string	enabled 172.18.255.71 192.168.1.1 addplus1 add +1 stripplus1 delete +1

## SBC configuration for Dual Trunk registration using G729 towards trunk

For this configuration, the SBC will need to transcode between G729 and PCMU codecs on the turnk and SFB side respectively. For transcoding, we configure the codec polices, to-trunk and to-sfb, mentioned below and apply them on the trunk and SFB realms respectively. The following configuration shows the configuration elements that need to be changed for the transcoding configuration to take effect

codec-policy	
name	to-trunk
allow-codecs	* PCMU:no G729::vadoff
add-codecs-on-egress	G729
order-codecs	
packetization-time	20
force-ptime	disabled
dtmf-in-audio	disabled
last-modified-by	admin@172.18.0.139
last-modified-date	2016-08-08 16:43:07
codec-policy	
name	to-sfb
allow-codecs	* G729:no
add-codecs-on-egress	PCMU
order-codecs	
packetization-time	20
force-ptime	disabled
dtmf-in-audio	disabled
last-modified-by	admin@172.18.0.139
last-modified-date	2016-08-08 16:43:07
media-profile	
name	G729

subname	vadoff
payload-type	18
parameters	annexb=no
playback-config	
name	transferg729
entry	
encoding	G729
filename	ringback-us-g729.raw
realm-config	
identifier	core
network-interfaces	s1p0:0
mm-in-realm	enabled
qos-enable	enabled
restricted-latching	sdp
codec-policy	core
realm-config	
identifier	ctl-trunk
network-interfaces	s0p0:0
mm-in-realm	enabled
qos-enable	enabled
media-policy	voip-default
codec-policy	access
realm-config	
identifier	core2
network-interfaces	s1p0:0
mm-in-realm	enabled
qos-enable	enabled
restricted-latching	sdp
codec-policy	core
realm-config	
identifier	ctl-trunk2
network-interfaces	s0p0:0
mm-in-realm	enabled
qos-enable	enabled
media-policy	voip-default
codec-policy	access

On the SFB server, add the ip address of the sip-interface of the core2 realm as a second PSTN gateway in the topology builder and also in the Routes tab of the Voice Routing section of the SFB Control Panel following the steps mentioned in the section – Configuring the Skype for Business server.

## Test Plan

Following is the test plan executed against this setup and results have been documented below

Title	Description	Test Setup	Status
Configure Dual Trunk on PBX	PBX is configured and connected to 2 PSTN GW/SBCs	The steps will be based on the type of PBX being utilized. Ensure that trunks are configured between PBX and SBC. Verify OPTIONS msgs from either PBX or SBC are being responded correctly by the other entity	Passed
Configure Dual Trunk on ITSP	ITSP is configured and connected to 2 PSTN GW/SBCs	The steps will be based on the type of SBC being utilized. Ensure the TWO SBCs are configured with individual trunks to ITSP	Passed
Regitration of Dual Trunks	Ensure that both trunks to ITSP are registered successfully using the individual trunk registration information	<ol> <li>Each SBC is configured with a trunk to ITSP and associated authentiation/digest and registration information.</li> <li>Invoke a command on SBC to register the trunk with ITSP.</li> <li>Verify that 200 OK is received from ITSP for both the trunks.</li> </ol>	Passed

Inhound DCTN colle nick	Varify that DCTN to DDV	1 Dial an inhound call to	
correct trunk to SBC	inbound calls arrive on	the PBX.	
	both the trunks when	2. Verify ringing is heard	
	multiple calls are made	by calling and called	
		parties.	
		3. Verify the trace shows	
		a valid ringing indication	
		message	
		4. Take called party	
		nhone off-hook	
		5 Verify that a media	
		nath is established in both	
		directions	
		C llang up colling portu	
			Dered
		7. Verify the IP/PBX	Passed
		receives a Bye message.	
		8. Make a note of the	
		Trunk on which the call	
		arrived to the SBC and	
		PBX.	
		9. Repeat the above	
		steps 3 more times (total	
		4 calls).	
		10. Verify that calls to	
		PBX arrive on both the	
		trunks.	
		11. Document Test	
		Results	
		12 Save Trace	
		IZ. SAVE HACE.	

PBX calls are delivered to PSTN on both the trunks	Calls from PBX to PSTN are delivered to ITSP/PSTN utilizing both the configured trunks	<ol> <li>Dial an outbound call from the PBX.</li> <li>Verify ringing is heard by calling and called parties.</li> <li>Verify the trace shows a valid ringing indication message</li> <li>Take called party phone off-hook.</li> <li>Verify that a media path established in both directions.</li> <li>Hang up Calling Party</li> <li>Verify the IP/PBX sends a Bye message.</li> <li>Make a note of the Trunk on which the call was sent to ITSP.</li> <li>Repeat the above steps 3 more times (total 4 calls).</li> <li>Verify that calls from PBX are sent out on both the trunks to ITSP.</li> <li>Verify each call has PAI sent per the trunk configuration</li> <li>Document Test Results.</li> <li>Save Trace.</li> </ol>	Passed
Alien TN calls on 1st trunk	Verify calls are successful with Alien TNs on 1st trunk	<ol> <li>After Alien TN is set up on a Trunk1 in</li> <li>CenturyLink Network</li> <li>PBX User Places a Call to PSTN</li> <li>PBX User receives a Call back</li> <li>PSTN receives ringing</li> <li>PSTN receives caller id of the Alien TN</li> <li>PSTN answers the call</li> <li>2 way audio is received</li> <li>PBX Phone releases</li> <li>Calls</li> <li>PSTN receives a Bye</li> </ol>	Passed

Alien TN calls on 2nd	Verify calls are successful	1. After Alien TN is set up	
trunk	with Alien TNs on 2nd	on a Trunk2 in	
	trunk	CenturyLink Network	
		2. PBX User Places a Call	
		to PSTN	
		3. PBX User receives ring	
		back	
		4. PSTN receives ringing	Passed
		5. PSTN receives caller id	
		of the Alien TN	
		6. PSTN answers the call	
		7. 2 way audio is received	
		8. PBX Phone releases	
		Calls	
		9. PSTN receives a Bye	
Failover of 1st trunk WAN	Ensure that calls are	1. Down the WAN	
- PSTN-PBX	delivered from PSTN to	interface associated with	
	PBX when the first trunk	Trunk 1.	
	has failed on the WAN	2. Make 3 calls from PSTN	
	side	to PBX (one after the	
		other)	
		3. Ensure that all 3 calls	
		are delivered to PBX	
		utilizing Trunk 2	Passed
		4. PBX user answers the	i assea
		call	
		5. Verify two way voice	
		path is established	
		6. Called Party hangs up	
		7. Both Calling and Called	
		parties are disconnected	
		8. Document results	
		9. Save traces	

Failover of 1st trunk WAN	Ensure that calls are	1. Down the WAN	
- PBX-PSTN	delivered from PBX to	interface associated with	
	PSTN when the first trunk	Trunk 1.	
	has failed on the WAN	2. Make 3 calls from PBX	
	side	to PSTN (one after the	
		other)	
		3. Ensure that all 3 calls	
		are delivered to PSTN	
		utilizing Trunk 2	Dessed
		4. PSTN user answers the	Passed
		call	
		5. Verify two way voice	
		path is established	
		6. Called Party hangs up	
		7. Both Calling and Called	
		parties are disconnected	
		8. Document results	
		9. Save traces	
Restore 1st trunk WAN:	Ensure that calls are	1. WAN interface	
PSTN-PBX	delivered from PSTN to	associated with Trunk 1 is	
	PBX when the first trunk	brought back into service.	
	has has been restored	2. Make 3 calls from PSTN	
		to PBX (one after the	
		other)	
		3. Ensure that at least one	
		call is delivered to the PBX	
		via Trunk 1	
		4. PBX user answers the	Passed
		call	
		5. Verify two way voice	
		path is established	
		6. Called Party hangs up	
		7. Both Calling and Called	
		parties are disconnected	
		8. Document results	
		9. Save traces	

Restore 1st trunk WAN: PBX-PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has has been restored	<ol> <li>WAN interface         <ul> <li>associated with Trunk 1 is</li> <li>brought back into service.</li> <li>Make 3 calls from PBX</li> <li>to PSTN (one after the             other)</li> <li>Ensure that at least one             call is delivered to the             PSTN via Trunk 1             </li> <li>PSTN user answers the             call             </li> <li>Verify two way voice             path is established             </li> <li>Called Party hangs up             </li> <li>Both Calling and Called             parties are disconnected             </li> <li>Document results             </li> </ul> </li> </ol>	Passed
Failover of 2nd trunk WAN: PSTN-PBX	Ensure that calls are delivered from PSTN to PBX when the second trunk has failed on the WAN side	<ol> <li>Down the WAN interface associated with Trunk 2.</li> <li>Make 3 calls from PSTN to PBX (one after the other)</li> <li>Ensure that all 3 calls are delivered to PBX utilizing Trunk 1</li> <li>PBX user answers the call</li> <li>Verify two way voice path is established</li> <li>Called Party hangs up</li> <li>Both Calling and Called parties are disconnected</li> <li>Document results</li> <li>Save traces</li> </ol>	Passed

Failover of 2nd trunk	Ensure that calls are	1. Down the WAN	
WAN: PBX-PSTN	delivered from PBX-PSTN	Interface associated with	
	when the second trunk	Trunk 2.	
	has failed on the WAN	2. Make 3 calls from PBX	
	side	to PSTN (one after the other)	
		3 Ensure that all 3 calls	
		are delivered to PBX	
		utilizing Trunk 1	
		4 PSTN user answers the	Passed
		call	
		5. Verify two way voice	
		path is established	
		6. Called Party hangs up	
		7. Both Calling and Called	
		parties are disconnected	
		8. Document results	
		9. Save traces	
Restore 2nd trunk WAN:	Ensure that calls are	1. WAN interface	
PSTN-PBX	delivered from PSTN to	associated with Trunk 2 is	
	PBX when the second	brought back into service.	
	trunk has has been	2. Make 3 calls from PSTN	
	restored	to PBX (one after the	
		other)	
		3. Ensure that at least one	
		call is delivered to the PBX	
		via Trunk 2	Passed
		4. PSTN user answers the	
		call	
		5. Verify two way voice	
		path is established	
		b. Called Party hangs up	
		7. Both Calling and Called	
		parties are disconnected	
		8. Document results	
		9. Save traces	

Restore 2nd trunk WAN: PBX-PSTN	Ensure that calls are delivered from PBX to PSTN when the second trunk has has been restored	<ol> <li>WAN interface         <ul> <li>associated with Trunk 2 is</li> <li>brought back into service.</li> <li>Make 3 calls from PBX</li> <li>to PSTN (one after the             other)</li> <li>Ensure that at least one             call is delivered to the PBX             via Trunk 2</li> <li>PBX user answers the             call</li> <li>Verify two way voice             path is established</li> <li>Called Party hangs up             <li>Both Calling and Called             parties are disconnected</li>             Socument results             </li> </ul> </li> </ol>	Passed
Failover of 1st trunk LAN - PBX to PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has failed on the LAN side	<ol> <li>Down the LAN interface associated with Trunk 1.</li> <li>Make 3 calls from PBX to PSTN (one after the other)</li> <li>Ensure that all 3 calls are delivered to PSTN utilizing Trunk 2</li> <li>PSTN user answers the call</li> <li>Verify two way voice path is established</li> <li>Called Party hangs up</li> <li>Both Calling and Called parties are disconnected</li> <li>Document results</li> <li>Save traces</li> </ol>	Passed

Failover of 1st trunk LAN - PSTN to PBX	Ensure that calls are delivered from PSTN to PBX when the first trunk has failed on the LAN side	<ol> <li>Down the LAN interface associated with Trunk 1.</li> <li>Make 3 calls from PSTN to PBX (one after the other)</li> <li>Ensure that all 3 calls are delivered to PBX utilizing Trunk 2</li> <li>PBX user answers the call</li> <li>Verify two way voice path is established</li> <li>Called Party hangs up</li> <li>Both Calling and Called parties are disconnected</li> <li>Document results</li> <li>Save traces</li> </ol>	Passed
Restore 1st trunk LAN - PBX to PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has has been restored	<ol> <li>LAN interface         <ul> <li>associated with Trunk 1 is</li> <li>brought back into service.</li> <li>Make 3 calls from PBX</li> <li>to PSTN (one after the other)</li> <li>Ensure that at least one call is delivered to the PSTN via Trunk 2</li> <li>PSTN user answers the call</li> <li>Verify two way voice path is established</li> <li>Called Party hangs up</li> <li>Both Calling and Called parties are disconnected</li> <li>Document results</li> <li>Save traces</li> </ul> </li> </ol>	Passed

Restore 1st trunk LAN - PSTN to PBX	Ensure that calls are delivered from PSTN to PBX when the first trunk has has been restored	<ol> <li>LAN interface         associated with Trunk 1 is         brought back into service.         Make 3 calls from PSTN             to PBX (one after the             other)         Ensure that at least one             call is delivered to the PBX             via Trunk 2          PBX user answers the             call          Service that a least one             call          Service that a least one             call is delivered to the PBX             via Trunk 2          A. PBX user answers the             call          Service the call          Service the call and called          Path calling and Called          parties are disconnected          8. Document results          9. Save traces      </li> </ol>	Passed
Failover of 2nd trunk LAN - PBX to PSTN	Ensure that calls are delivered from PBX to PSTN when the second trunk has failed on the LAN side	<ol> <li>Down the LAN interface associated with Trunk 2.</li> <li>Make 3 calls from PBX to PSTN (one after the other)</li> <li>Ensure that all 3 calls are delivered to PSTN utilizing Trunk 1</li> <li>PSTN user answers the call</li> <li>Verify two way voice path is established</li> <li>Called Party hangs up</li> <li>Both Calling and Called parties are disconnected</li> <li>Document results</li> <li>Save traces</li> </ol>	Passed

Failover of 2nd trunk LAN - PSTN to PBX	Ensure that calls are delivered from PSTN to PBX when the second trunk has failed on the LAN side	<ol> <li>Down the LAN interface associated with Trunk 2.</li> <li>Make 3 calls from PSTN to PBX (one after the other)</li> <li>Ensure that all 3 calls are delivered to PBX utilizing Trunk 1</li> <li>PBX user answers the call</li> <li>Verify two way voice path is established</li> <li>Called Party hangs up</li> <li>Both Calling and Called parties are disconnected</li> <li>Document results</li> <li>Save traces</li> </ol>	Passed
Restore 2nd trunk LAN - PBX to PSTN	Ensure that calls are delivered from PBX to PSTN when the second trunk has has been restored	<ol> <li>LAN interface         <ul> <li>associated with Trunk 2 is</li> <li>brought back into service.</li> <li>Make 3 calls from PBX</li> <li>to PSTN (one after the other)</li> <li>Ensure that at least one call is delivered to the PSTN via Trunk 2</li> <li>PSTN user answers the call</li> <li>Verify two way voice path is established</li> <li>Called Party hangs up</li> <li>Both Calling and Called parties are disconnected</li> <li>Document results</li> <li>Save traces</li> </ul> </li> </ol>	Passed

Restore 2nd trunk LAN - PSTN to PBX	Ensure that calls are delivered from PSTN to PBX when the second trunk has has been restored	<ol> <li>LAN interface         <ul> <li>associated with Trunk 2 is</li> <li>brought back into service.</li> <li>Make 3 calls from PSTN</li> <li>to PBX (one after the             other)</li> <li>Ensure that at least one             call is delivered to the PBX             via Trunk 2</li> <li>PBX user answers the             call</li> <li>Verify two way voice             path is established</li> <li>Called Party hangs up             </li> </ul> </li> </ol>	Passed
Configure Dual Trunk on PBX	PBX is configured and connected to 2 PSTN GW/SBCs	8. Document results 9. Save traces The steps will be based on the type of PBX being utilized. Ensure that trunks are configured between PBX and SBC. Verify OPTIONS msgs from either PBX or SBC are being responded correctly by the other entity	Passed
Configure Dual Trunk on ITSP	ITSP is configured and connected to 2 PSTN GW/SBCs	The steps will be based on the type of SBC being utilized. Ensure the TWO SBCs are configured with individual trunks to ITSP	Passed
Regitration of Dual Trunks	Ensure that both trunks to ITSP are registered successfully using the individual trunk registration information	<ol> <li>Each SBC is configured with a trunk to ITSP and associated authentiation/digest and registration information.</li> <li>Invoke a command on SBC to register the trunk with ITSP.</li> <li>Verify that 200 OK is received from ITSP for both the trunks.</li> </ol>	Passed

Inhound DCTN collemint	Varify that DCTN to DDV	1 Dial an inhound call to	
Indound PSTN calls pick	Verify that PSTN to PBX	1. Diai an indound call to	
correct trunk to SBC	inbound calls arrive on	the PBX.	
	both the trunks when	2. Verify ringing is heard	
	multiple calls are made	by calling and called	
		parties.	
		3. Verify the trace shows	
		a valid ringing indication	
		message	
		4. Take called party	
		nhone off-hook	
		5 Verify that a media	
		nath is established in both	
		directions	
		G Hang up calling party	
			Dered
		7. Verify the IP/PBX	Passed
		receives a Bye message.	
		8. Make a note of the	
		Trunk on which the call	
		arrived to the SBC and	
		PBX.	
		9. Repeat the above	
		steps 3 more times (total	
		4 calls).	
		10. Verify that calls to	
		PBX arrive on both the	
		trunks.	
		11. Document Test	
		Results.	
		12. Save Trace.	

PBX calls are delivered to PSTN on both the trunks	Calls from PBX to PSTN are delivered to ITSP/PSTN utilizing both the configured trunks	<ol> <li>Dial an outbound call from the PBX.</li> <li>Verify ringing is heard by calling and called parties.</li> <li>Verify the trace shows a valid ringing indication message</li> <li>Take called party phone off-hook.</li> <li>Verify that a media path established in both directions.</li> <li>Hang up Calling Party</li> <li>Verify the IP/PBX sends a Bye message.</li> <li>Make a note of the Trunk on which the call was sent to ITSP.</li> <li>Repeat the above steps 3 more times (total 4 calls).</li> <li>Verify that calls from PBX are sent out on both the trunks to ITSP.</li> <li>Verify each call has PAI sent per the trunk configuration</li> <li>Document Test Results.</li> <li>Save Trace.</li> </ol>	Passed
Alien TN calls on 1st trunk	Verity calls are successful with Alien TNs on 1st trunk	<ol> <li>After Alien TN is set up on a Trunk1 in</li> <li>CenturyLink Network</li> <li>PBX User Places a Call to PSTN</li> <li>PBX User receives a Call back</li> <li>PSTN receives ringing</li> <li>PSTN receives caller id of the Alien TN</li> <li>PSTN answers the call</li> <li>2 way audio is received</li> <li>PBX Phone releases</li> <li>Calls</li> <li>PSTN receives a Bye</li> </ol>	Passed

Alien TN calls on 2nd	Verify calls are successful	1. After Alien TN is set up	
trunk	with Alien TNs on 2nd	on a Trunk2 in	
	trunk	CenturyLink Network	
		2. PBX User Places a Call	
		to PSTN	
		3. PBX User receives ring	
		back	
		4. PSTN receives ringing	Passed
		5. PSTN receives caller id	
		of the Alien TN	
		6. PSTN answers the call	
		7. 2 way audio is received	
		8. PBX Phone releases	
		Calls	
		9. PSTN receives a Bye	
Failover of 1st trunk WAN	Ensure that calls are	1. Down the WAN	
- PSTN-PBX	delivered from PSTN to	interface associated with	
	PBX when the first trunk	Trunk 1.	
	has failed on the WAN	2. Make 3 calls from PSTN	
	side	to PBX (one after the	
		other)	
		3. Ensure that all 3 calls	
		are delivered to PBX	
		utilizing Trunk 2	Passed
		4. PBX user answers the	1 usseu
		call	
		5. Verify two way voice	
		path is established	
		6. Called Party hangs up	
		7. Both Calling and Called	
		parties are disconnected	
		8. Document results	
		9. Save traces	

Failover of 1st trunk WAN	Ensure that calls are	1. Down the WAN	
- PBX-PSTN	delivered from PBX to	interface associated with	
	PSTN when the first trunk	Trunk 1.	
	has failed on the WAN	2. Make 3 calls from PBX	
	side	to PSTN (one after the	
		other)	
		3. Ensure that all 3 calls	
		are delivered to PSTN	
		utilizing Trunk 2	Dessed
		4. PSTN user answers the	Passed
		call	
		5. Verify two way voice	
		path is established	
		6. Called Party hangs up	
		7. Both Calling and Called	
		parties are disconnected	
		8. Document results	
		9. Save traces	
Restore 1st trunk WAN:	Ensure that calls are	1. WAN interface	
PSTN-PBX	delivered from PSTN to	associated with Trunk 1 is	
	PBX when the first trunk	brought back into service.	
	has has been restored	2. Make 3 calls from PSTN	
		to PBX (one after the	
		other)	
		3. Ensure that at least one	
		call is delivered to the PBX	
		via Trunk 1	- ·
		4. PBX user answers the	Passed
		call	
		5. Verify two way voice	
		path is established	
		6. Called Party hangs up	
		7. Both Calling and Called	
		parties are disconnected	
		8. Document results	
		9. Save traces	

Restore 1st trunk WAN: PBX-PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has has been restored	<ol> <li>WAN interface         associated with Trunk 1 is         brought back into service.         Make 3 calls from PBX         to PSTN (one after the         other)         Ensure that at least one         call is delivered to the         PSTN via Trunk 1         PSTN user answers the         call         Verify two way voice         path is established         </li> </ol>	Passed
		<ul><li>6. Called Party hangs up</li><li>7. Both Calling and Called parties are disconnected</li><li>8. Document results</li><li>9. Save traces</li></ul>	
Failover of 2nd trunk WAN: PSTN-PBX	Ensure that calls are delivered from PSTN to PBX when the second trunk has failed on the WAN side	<ol> <li>Down the WAN interface associated with Trunk 2.</li> <li>Make 3 calls from PSTN to PBX (one after the other)</li> <li>Ensure that all 3 calls are delivered to PBX utilizing Trunk 1</li> <li>PBX user answers the call</li> <li>Verify two way voice path is established</li> <li>Called Party hangs up</li> <li>Both Calling and Called parties are disconnected</li> <li>Document results</li> <li>Save traces</li> </ol>	Passed

Failover of 2nd trunk	Ensure that calls are	1. Down the WAN	
WAN: PBX-PSTN	delivered from PBX-PSTN	interface associated with	
	when the second trunk	Trunk 2.	
	has failed on the WAN	2. Make 3 calls from PBX	
	side	to PSTN (one after the	
		other)	
		3. Ensure that all 3 calls	
		are delivered to PBX	
		utilizing Trunk 1	Dassad
		4. PSTN user answers the	Passeu
		call	
		5. Verify two way voice	
		path is established	
		6. Called Party hangs up	
		7. Both Calling and Called	
		parties are disconnected	
		8. Document results	
		9. Save traces	
Restore 2nd trunk WAN:	Ensure that calls are	1. WAN interface	
PSTN-PBX	delivered from PSTN to	associated with Trunk 2 is	
	PBX when the second	brought back into service.	
	trunk has has been	2. Make 3 calls from PSTN	
	restored	to PBX (one after the	
		other)	
		3. Ensure that at least one	
		call is delivered to the PBX	
		via Trunk 2	Description
		4. PSTN user answers the	Passed
		call	
		5. Verify two way voice	
		path is established	
		6. Called Party hangs up	
		7. Both Calling and Called	
		parties are disconnected	
		8. Document results	
		9. Save traces	

Restore 2nd trunk WAN: PBX-PSTN	Ensure that calls are delivered from PBX to PSTN when the second trunk has has been restored	<ol> <li>WAN interface         <ul> <li>associated with Trunk 2 is</li> <li>brought back into service.</li> <li>Make 3 calls from PBX</li> <li>to PSTN (one after the             other)</li> <li>Ensure that at least one             call is delivered to the PBX             via Trunk 2</li> <li>PBX user answers the             call</li> <li>Verify two way voice             path is established             <li>Called Party hangs up             </li> <li>Both Calling and Called             parties are disconnected             </li> <li>Document results             </li> </li></ul> </li> </ol>	Passed
Failover of 1st trunk LAN - PBX to PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has failed on the LAN side	<ol> <li>Down the LAN interface associated with Trunk 1.</li> <li>Make 3 calls from PBX to PSTN (one after the other)</li> <li>Ensure that all 3 calls are delivered to PSTN utilizing Trunk 2</li> <li>PSTN user answers the call</li> <li>Verify two way voice path is established</li> <li>Called Party hangs up</li> <li>Both Calling and Called parties are disconnected</li> <li>Document results</li> <li>Save traces</li> </ol>	Passed

Failover of 1st trunk LAN - PSTN to PBX	Ensure that calls are delivered from PSTN to PBX when the first trunk has failed on the LAN side	<ol> <li>Down the LAN interface associated with Trunk 1.</li> <li>Make 3 calls from PSTN to PBX (one after the other)</li> <li>Ensure that all 3 calls are delivered to PBX utilizing Trunk 2</li> <li>PBX user answers the call</li> <li>Verify two way voice path is established</li> <li>Called Party hangs up</li> <li>Both Calling and Called parties are disconnected</li> <li>Document results</li> <li>Save traces</li> </ol>	Passed
Restore 1st trunk LAN - PBX to PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has has been restored	<ol> <li>LAN interface         <ul> <li>associated with Trunk 1 is</li> <li>brought back into service.</li> <li>Make 3 calls from PBX</li> <li>to PSTN (one after the other)</li> <li>Ensure that at least one call is delivered to the PSTN via Trunk 2</li> <li>PSTN user answers the call</li> <li>Verify two way voice path is established</li> <li>Called Party hangs up</li> <li>Both Calling and Called parties are disconnected</li> <li>Document results</li> <li>Save traces</li> </ul> </li> </ol>	Passed

Restore 1st trunk LAN - PSTN to PBX	Ensure that calls are delivered from PSTN to PBX when the first trunk has has been restored	<ol> <li>LAN interface         associated with Trunk 1 is         brought back into service.         Make 3 calls from PSTN         to PBX (one after the         other)         Ensure that at least one         call is delivered to the PBX         via Trunk 2         PBX user answers the         call         Verify two way voice         path is established         Called Party hangs up         Both Calling and Called         parties are disconnected         Bocument results         Save traces</li></ol>	Passed
Failover of 2nd trunk LAN - PBX to PSTN	Ensure that calls are delivered from PBX to PSTN when the second trunk has failed on the LAN side	<ol> <li>Down the LAN interface associated with Trunk 2.</li> <li>Make 3 calls from PBX to PSTN (one after the other)</li> <li>Ensure that all 3 calls are delivered to PSTN utilizing Trunk 1</li> <li>PSTN user answers the call</li> <li>Verify two way voice path is established</li> <li>Called Party hangs up</li> <li>Both Calling and Called parties are disconnected</li> <li>Document results</li> <li>Save traces</li> </ol>	Passed

Failover of 2nd trunk LAN - PSTN to PBX	Ensure that calls are delivered from PSTN to PBX when the second trunk has failed on the LAN side	<ol> <li>Down the LAN interface associated with Trunk 2.</li> <li>Make 3 calls from PSTN to PBX (one after the other)</li> <li>Ensure that all 3 calls are delivered to PBX utilizing Trunk 1</li> <li>PBX user answers the call</li> <li>Verify two way voice path is established</li> <li>Called Party hangs up</li> <li>Both Calling and Called parties are disconnected</li> <li>Document results</li> <li>Save traces</li> </ol>	Passed
Restore 2nd trunk LAN - PBX to PSTN	Ensure that calls are delivered from PBX to PSTN when the second trunk has has been restored	<ol> <li>LAN interface         associated with Trunk 2 is         brought back into service.         Make 3 calls from PBX         to PSTN (one after the         other)         Ensure that at least one         call is delivered to the         PSTN via Trunk 2         PSTN via Trunk 2         PSTN user answers the         call         Call State         Call A State         Constant         Constant         Source         Save traces         Save traces<!--</td--><td>Passed</td></li></ol>	Passed

7. Both Calling and Called parties are disconnected 8. Document results	Restore 2nd trunk LAN - PSTN to PBX	Ensure that calls are delivered from PSTN to PBX when the second trunk has has been restored	<ol> <li>LAN interface         associated with Trunk 2 is         brought back into service.         Make 3 calls from PSTN             to PBX (one after the             other)          Sensure that at least one             call is delivered to the PBX             via Trunk 2          PBX user answers the             call          Sensure that at least one call is delivered to the PBX             via Trunk 2          Sensure that at least one call is delivered to the PBX          via Trunk 2         Sensure that at least one call          Sensure that at least one call is delivered to the PBX      </li> </ol>	Passed	
9. Save traces			9. Save traces		

.

## **Troubleshooting Tools**

If you find that you are not able to complete calls or have problems with the test cases, there are a few tools available for Windows Server, Lync Server, and the Oracle SBC like logging and tracing which may be of assistance. In this section we will provide a list of tools which you can use to aid in troubleshooting any issues you may encounter.

Since we are concerned with communication between the Lync Server mediation server and the SBC we will focus on the troubleshooting tools to use between those devices if calls are not working or tests are not passing.

## Microsoft Network Monitor (NetMon)

NetMon is a network protocol analyzer which is freely downloadable from Microsoft. It can be found at <u>www.microsoft.com/downloads</u>. NetMon could be installed on the Lync Server mediation server, the Lync Server Standard Edition server, or Enterprise Edition front end server.

#### Wireshark

Wireshark is also a network protocol analyzer which is freely downloadable from <u>www.wireshark.org</u>. Wireshark could be installed on the Lync Server mediation server, the Lync Server Standard Edition server, or MCS Enterprise Edition front end server.

#### **Eventviewer**

There are several locations in the event viewer where you can find valuable information to aid in troubleshooting issues with your deployment.

With the requirement that there is a completely functioning Lync Server with Enterprise Voice deployment in place, there are only a few areas in which one would use the Event Viewer for troubleshooting:

- The Enterprise Voice client;
- The Lync Server Front End server;
- A Lync Server Standard Edition Server; and
- A Lync Server Mediation Server.

#### On the Oracle SBC 4600 Series

The Oracle SBC provides a rich set of statistical counters available from the ACLI, as well as log file output with configurable detail. The follow sections detail enabling, adjusting and accessing those interfaces.

Resetting the statistical counters, enabling logging and restarting the log files.

At the SBC Console:

```
oraclesbc1# reset sipd
oraclesbc1# notify sipd debug
oraclesbc1#
enabled SIP Debugging
oraclesbc1# notify all rotate-logs
```

Examining the log files

Note: You will FTP to the management interface of the SBC with the username user and user mode password (the default is "acme").

```
C:\Documents and Settings\user>ftp 192.168.5.24
Connected to 192.168.85.55.
220 oraclesbc1FTP server (VxWorks 6.4) ready.
User (192.168.85.55: (none)): user
331 Password required for user.
Password: acme
230 User user logged in.
ftp> cd /ramdrv/logs
250 CWD command successful.
ftp> get sipmsg.log
200 PORT command successful.
150 Opening ASCII mode data connection for '/ramdrv/logs/sipmsg.log' (3353
bytes).
226 Transfer complete.
ftp: 3447 bytes received in 0.00Seconds 3447000.00Kbytes/sec.
ftp> get log.sipd
200 PORT command successful.
150 Opening ASCII mode data connection for '/ramdrv/logs/log.sipd' (204681
bytes).
226 Transfer complete.
ftp: 206823 bytes received in 0.11Seconds 1897.46Kbytes/sec.
ftp> bye
221 Goodbye.
```

You may now examine the log files with the text editor of your choice.

Through the Web GUI

You can also check the display results of filtered SIP session data from the Oracle Enterprise Session Border Controller, and provides traces in a common log format for local viewing or for exporting to your PC. Please check the "Monitor and Trace" section (page 145) of the Web GUI User Guide available at <a href="http://docs.oracle.com/cd/E56581\_01/index.htm">http://docs.oracle.com/cd/E56581\_01/index.htm</a>

#### Telnet

Since we are working within an architecture which uses bound TCP listening ports for functionality, the simplest form of troubleshooting can be seeing if the devices are listening on a particular port, as well as confirming that the there is nothing blocking them such as firewalls. Ensure that you have a TELNET client available on a workstation as well as on the Lync Server mediation server.

The Lync Server mediation server will listen on TCP port 5067 by default for SIP signaling. In our example we are listening on 5060 on the PSTN facing NIC. From the Standard Edition pool or Enterprise Edition pool the Mediation Server would be listening on port 5061. Tests may include:

- Client to pool server: telnet <servername> 5061
- Pool server to Mediation Server: telnet <servername> 5061

## On the Lync Server

## Lync Server Logging Tool

The Skype for Business Logging Tool provides internal traces and messaging between different Skype for Business elements like Front-end, Mediation server, Lync Clients, etc. File name is OCSReskit.msi. Once installed, it can be accessed from any one of the Lync Server servers by running Start/Microsoft Skype for Business/Lync Server Logging Tool.

R	Lync Server 2013 Log	ging Tool		-		x
Logging Options  Components  Client VersionFilter  CLSQent CLSController  CLSController  CLSControllerLib  CLSControllerLib  CLSControllerLib  CLSControllerLib  CLSControllerLib  CLSControllerLib  CLSControllerLib  CLSControllerLib  CLSControllerLib  CLSCONTRO  DataMCU DataMCU DataMCU DataMCU DataMCUURunTime DeviceUpdateHttpHandler DataMCU DataMCU DataMCUG BexumRouting HybridConfig IIMFilter ImMcu InboundRouting IncomingFederation Infrastructure V	Level Fatal Errors Errors Warnings Information Verbose All Elags TF_COMPONENT TF_COMPONENT TF_PROTOCOL TF_DIAG All Flags	Global Options Log File Options Type Circular Sequential New File Real Time Options Filter Options Filter Options Enabled Include Filters Exclude Filters	Maximum Size 20 Append to Display or Edit	e: ] MB b log file nly Clea	r	
Log File Folder: C:\Windows\Tracin	ng	Adversed Outing	5.4	B	rowse	
Start Logging View Log Files	Analyze Log Files	Advanced Options	Exit	He	elp	
No active log session. Check the components you wish to log Start Logging button to start logging the checked components of	in the list on the left. For each ch with the configured level and flags.	ecked component, configure the log le	vel and flags for that (	compone	nt. Ci	lick

# Appendix A

## Accessing the ACLI

Access to the ACLI is provided by:

- The serial console connection;
- TELNET, which is enabled by default but may be disabled; and
- SSH, this must be explicitly configured.

Initial connectivity will be through the serial console port. At a minimum, this is how to configure the management (eth0) interface on the SBC.



## **ACLI Basics**

There are two password protected modes of operation within the ACLI, User mode and Superuser mode.

When you establish a connection to the SBC, the prompt for the User mode password appears. The default password is acme.

User mode consists of a restricted set of basic monitoring commands and is identified by the greater than sign (>) in the system prompt after the target name. You cannot perform configuration and maintenance from this mode.



The Superuser mode allows for access to all system commands for operation, maintenance, and administration. This mode is identified by the pound sign (#) in the prompt after the target name. To enter the Superuser mode, issue the enable command in the User mode.



From the Superuser mode, you can perform monitoring and administrative tasks; however you cannot configure any elements. To return to User mode, issue the exit command.

You must enter the Configuration mode to configure elements. For example, you can access the configuration branches and configuration elements for signaling and media configurations. To enter the Configuration mode, issue the configure terminal command in the Superuser mode.

Configuration mode is identified by the word configure in parenthesis followed by the pound sign (#) in the prompt after the target name, for example, **oraclesbc1(configure)#**. To return to the Superuser mode, issue the **exit** command.



In the configuration mode, there are six configuration branches:

- bootparam;
- ntp-sync;
- media-manager;
- session-router;
- system; and
- security.



- boot device The global management port, usually eth0
- file name The boot path and the image file.
- inet on ethernet The IP address and subnet mask (in hex) of the management port of the SD.
- host inet -The IP address of external server where image file resides.
- user and ftp password Used to boot from the external FTP server.
- gateway inet The gateway IP address for reaching the external server, if the server is located in a different network.

```
'.' = clear field; '-' = go to previous field; q = quit
                       : eth0
processor number
boot device
                      : 0
host name
                       .
file name
                      : /tffs0/nnSCX620.gz
inet on ethernet (e) : 10.0.3.11:ffff0000
inet on backplane (b)
                       .
                       : 10.0.3.100
host inet (h)
host inet (h)
gateway inet (g)
user (u)
                      : 10.0.0.1
                       : anonymous
user (u)
ftp password (pw) (blank = rsh) : anonymous
             : 0x8
flags (f)
target name (tn)
startup script (s)
                      : MCS14-IOT-SD
                      .
other (o)
```

The ntp-sync branch provides access to ntp server configuration commands for synchronizing the SBC time and date.

The security branch provides access to security configuration.

The system branch provides access to basic configuration elements as system-config, snmp-community, redundancy, physical interfaces, network interfaces, etc.

The session-router branch provides access to signaling and routing related elements, including H323-config, sip-config, iwf-config, local-policy, sip-manipulation, session-agent, etc.

The media-manager branch provides access to media-related elements, including realms, steering pools, dns-config, mediamanager, and so forth.

You will use media-manager, session-router, and system branches for most of your working configuration.

#### **Configuration Elements**

The configuration branches contain the configuration elements. Each configurable object is referred to as an element. Each element consists of a number of configurable parameters.

Some elements are single-instance elements, meaning that there is only one of that type of the element - for example, the global system configuration and redundancy configuration.

Some elements are multiple-instance elements. There may be one or more of the elements of any given type. For example, physical and network interfaces.

Some elements (both single and multiple instance) have sub-elements. For example:

- SIP-ports are children of the sip-interface element
- peers are children of the redundancy element
- destinations are children of the peer element

#### **Creating an Element**
- 1. To create a single-instance element, you go to the appropriate level in the ACLI path and enter its parameters. There is no need to specify a unique identifier property because a single-instance element is a global element and there is only one instance of this element.
- 2. When creating a multiple-instance element, you must specify a unique identifier for each instance of the element.
- 3. It is important to check the parameters of the element you are configuring before committing the changes. You do this by issuing the **show** command before issuing the **done** command. The parameters that you did not configure are filled with either default values or left empty.
- 4. On completion, you must issue the **done** command. The done command causes the configuration to be echoed to the screen and commits the changes to the volatile memory. It is a good idea to review this output to ensure that your configurations are correct.
- 5. Issue the **exit** command to exit the selected element.

Note that the configurations at this point are not permanently saved yet. If the SBC reboots, your configurations will be lost.

## **Editing an Element**

The procedure of editing an element is similar to creating an element, except that you must select the element that you will edit before editing it.

- 1. Enter the element that you will edit at the correct level of the ACLI path.
- Select the element that you will edit, and view it before editing it. The select command loads the element to the volatile memory for editing. The show command allows you to view the element to ensure that it is the right one that you want to edit.
- 3. Once you are sure that the element you selected is the right one for editing, edit the parameter one by one. The new value you provide will overwrite the old value.
- 4. It is important to check the properties of the element you are configuring before committing it to the volatile memory. You do this by issuing the **show** command before issuing the **done** command.
- 5. On completion, you must issue the **done** command.
- 6. Issue the exit command to exit the selected element.

Note that the configurations at this point are not permanently saved yet. If the SBC reboots, your configurations will be lost.

### **Deleting an Element**

The **no** command deletes an element from the configuration in editing.

To delete a single-instance element,

- 1. Enter the **no** command from within the path for that specific element
- 2. Issue the **exit** command.

To delete a multiple-instance element,

- 1. Enter the **no** command from within the path for that particular element. The key field prompt, such as <name>:<sub-port-id>, appears.
- 2. Use the <Enter> key to display a list of the existing configured elements.
- 3. Enter the number corresponding to the element you wish to delete.
- 4. Issue the select command to view the list of elements to confirm that the element was removed.

Note that the configuration changes at this point are not permanently saved yet. If the SBC reboots, your configurations will be lost.

### **Configuration Versions**

At any time, three versions of the configuration can exist on the SBC: the edited configuration, the saved configuration, and the running configuration.

- The edited configuration this is the version that you are making changes to. This version of the configuration is stored in the SBC's volatile memory and will be lost on a reboot.
   To view the editing configuration, issue the show configuration command.
- The **saved configuration** on issuing the **save-config** command, the edited configuration is copied into the nonvolatile memory on the SBC and becomes the saved configuration. Because the saved configuration has not been activated yet, the changes in the configuration will not take effect. On reboot, the last activated configuration (i.e., the last running configuration) will be loaded, not the saved configuration.
- The **running configuration** is the saved then activated configuration. On issuing the **activate-config** command, the saved configuration is copied from the non-volatile memory to the volatile memory. The saved configuration is activated and becomes the running configuration. Although most of the configurations can take effect once being activated without reboot, some configurations require a reboot for the changes to take effect. To view the running configuration, issue command show **running-config**.

# Saving the Configuration

The **save-config** command stores the edited configuration persistently.

Because the saved configuration has not been activated yet, changes in configuration will not take effect. On reboot, the last activated configuration (i.e., the last running configuration) will be loaded. At this stage, the saved configuration is different from the running configuration.

Because the saved configuration is stored in non-volatile memory, it can be accessed and activated at later time.

Upon issuing the **save-config** command, the SBC displays a reminder on screen stating that you must use the **activate-config** command if you want the configurations to be updated.

```
oraclesbc1 # save-config
Save-Config received, processing.
waiting 1200 for request to finish
Request to 'SAVE-CONFIG' has Finished,
Save complete
Currently active and saved configurations do not match!
To sync & activate, run 'activate-config' or 'reboot activate'.
oraclesbc1 #
```

## Activating the Configuration

On issuing the **activate-config** command, the saved configuration is copied from the non-volatile memory to the volatile memory. The saved configuration is activated and becomes the running configuration.

Some configuration changes are service affecting when activated. For these configurations, the SBC warns that the change could have an impact on service with the configuration elements that will potentially be service affecting. You may decide whether or not to continue with applying these changes immediately or to apply them at a later time.

oraclesbcl# activate-config Activate-Config received, processing. waiting 120000 for request to finish Request to 'ACTIVATE-CONFIG' has Finished, Activate Complete oraclesbcl#



CONNECT WITH US

blogs.oracle.com/oracle facebook.com/oracle witter.com/oracle

oracle.com

# Integrated Cloud Applications & Platform Services

**Oracle Corporation, World Headquarters** 

Redwood Shores, CA 94065, USA

500 Oracle Parkway

Copyright © 2015, Oracle and/or its affiliates. All rights reserved. This document is provided for information purposes only, and the contents hereof are subject to change without notice. This document is not warranted to be error-free, nor subject to any other warranties or conditions, whether expressed orally or implied in law, including implied warranties and conditions of merchantability or fitness for a particular purpose. We specifically disclaim any liability with respect to this document, and no contractual obligations are formed either directly or indirectly by this document. This document may not be reproduced or transmitted in any form or by any means, electronic or mechanical, for any purpose, without our prior written permission.

Worldwide Inquiries Phone: +1.650.506.7000

Fax: +1.650.506.7200

Oracle and Java are registered trademarks of Oracle and/or its affiliates. Other names may be trademarks of their respective owners

Intel and Intel Xeon are trademarks or registered trademarks of Intel Corporation. All SPARC trademarks are used under license and are trademarks or registered trademarks of SPARC International, Inc. AMD, Opteron, the AMD logo, and the AMD Opteron logo are trademarks or registered trademarks of Advanced Micro Devices. UNIX is a registered trademark of The Open Group. 0615