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

Technical Application Note

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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

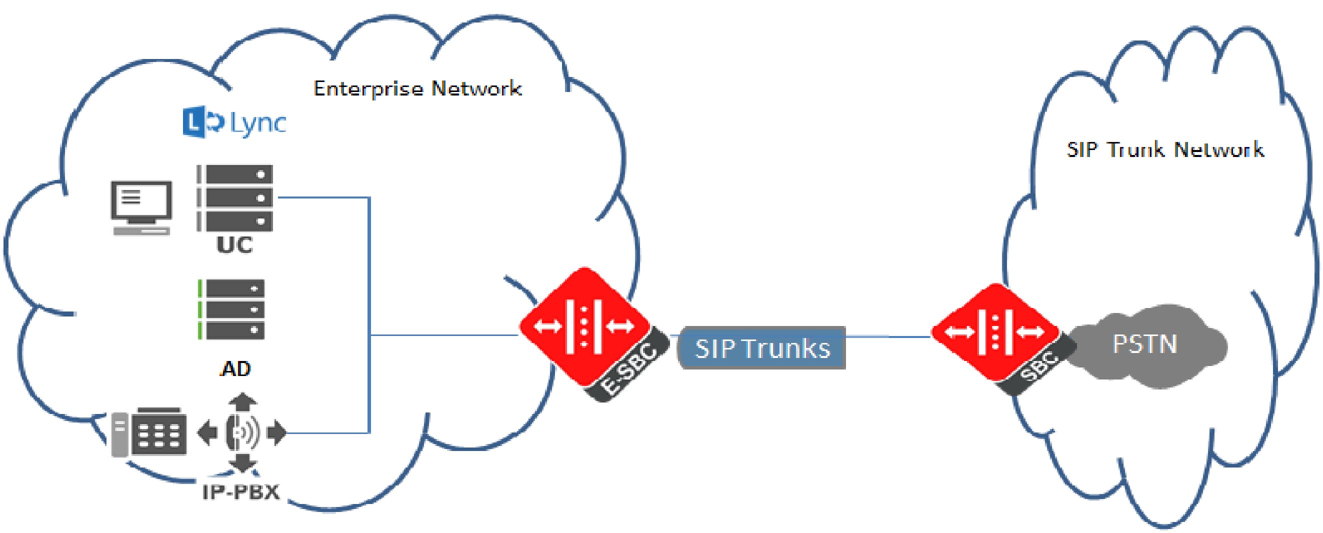
This is a technical document intended for telecommunications engineers with the purpose of configuring the Oracle Enterprise SBC and the Skype for Business Server. There will be steps that require navigating Microsoft Windows Server as well as the Acme Packet Command Line Interface (ACLI). Understanding 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.

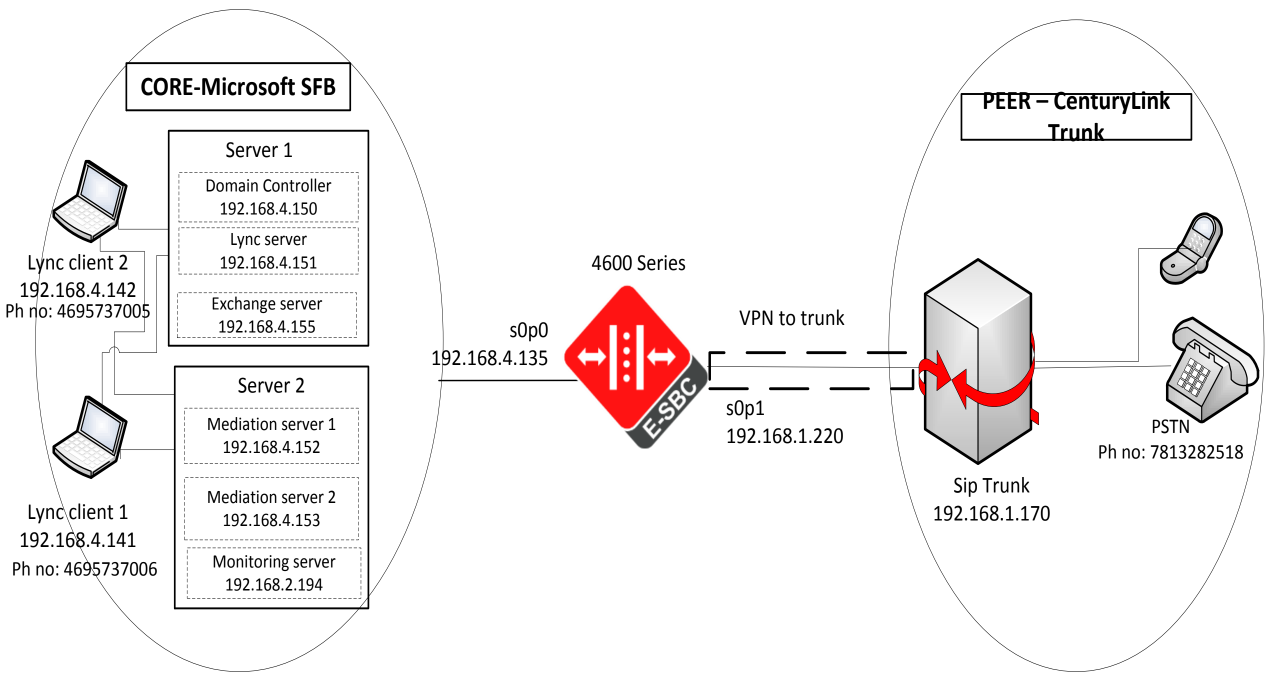
As per the CenturyLink network requirements, the customer devices are required to register and support authentication. The Oracle ESBC supports authentication and performs registration on behalf of the Skype for Business Server. The registration can be done using either the single trunk architecture 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:

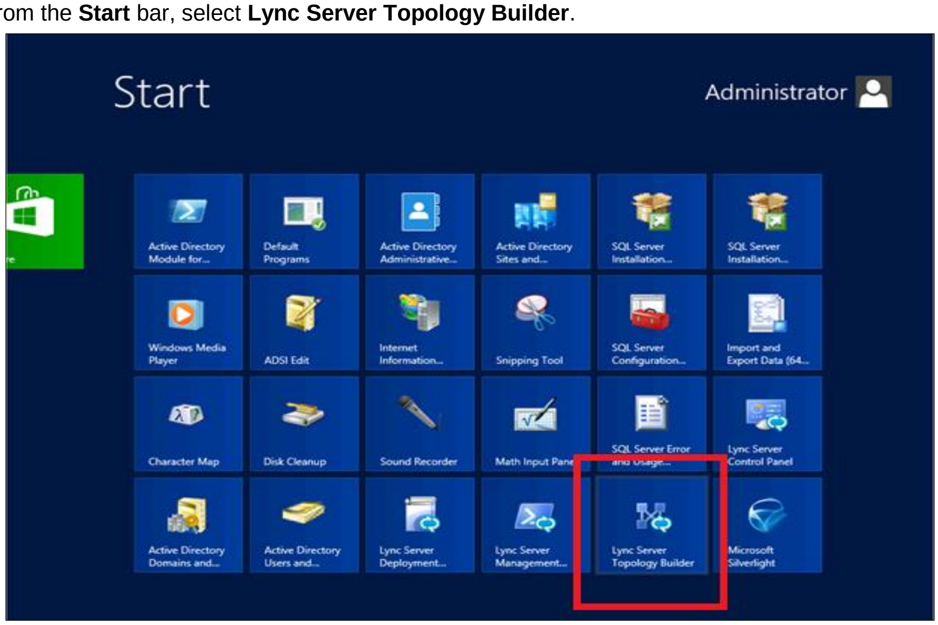
- IP addresses of the external facing NICs of the Mediation Servers
- IP address of the sip interface of the SBC facing the Mediation servers
- Rights to administer Lync Server Topology Builder
- Access to the Lync Server Topology Builder

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.

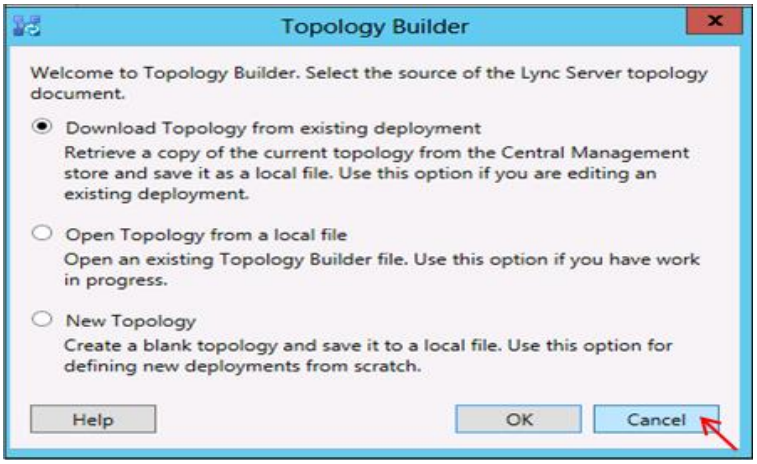
Adding the SBC as a PSTN gateway

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

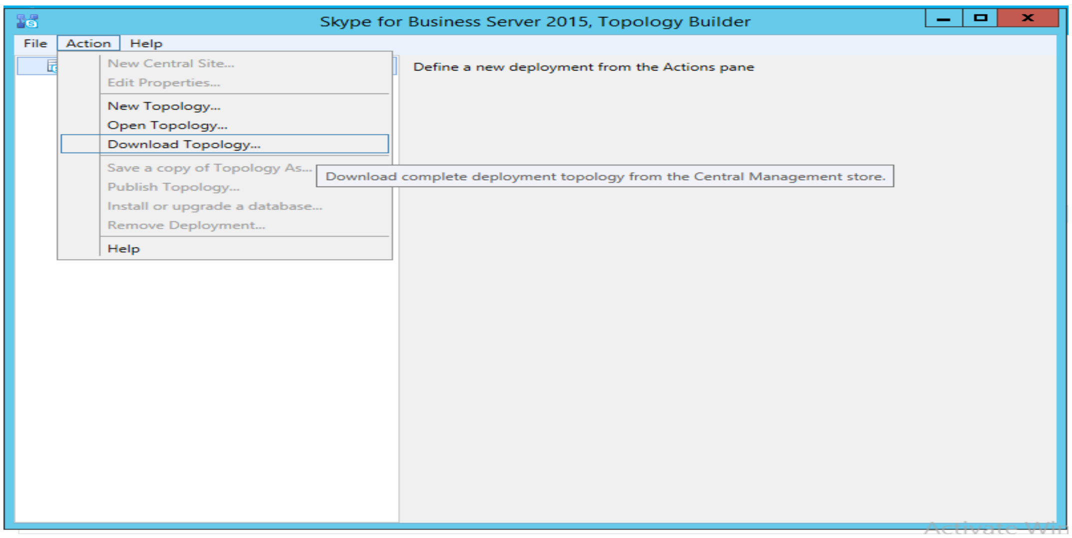
1. On the server where the Topology Builder is located, start the console.
2. From the **Start** bar, select **Lync Server Topology Builder**.



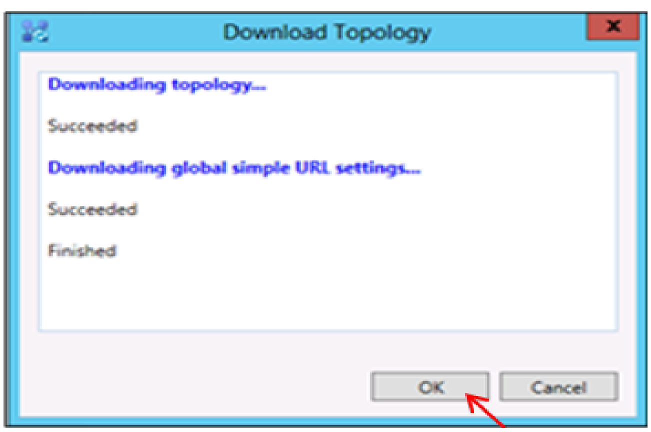
3. The opening screen of the Topology builder will be displayed. Click on the **Cancel** button.



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

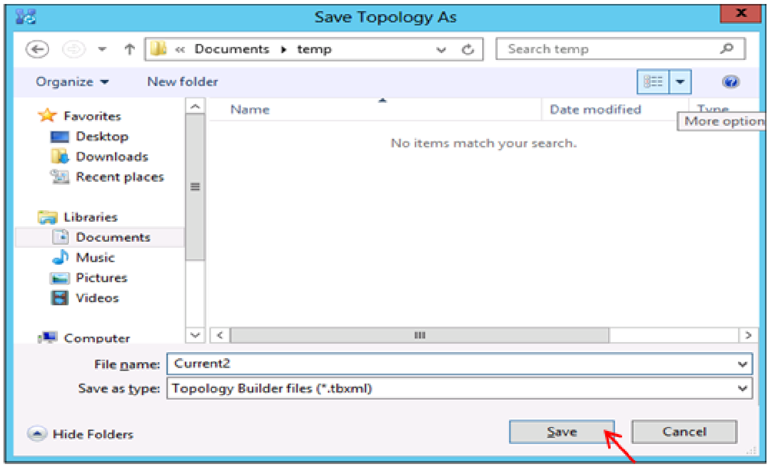


5. You will then see a screen showing that you have successfully imported the topology. Click the **Ok** button.

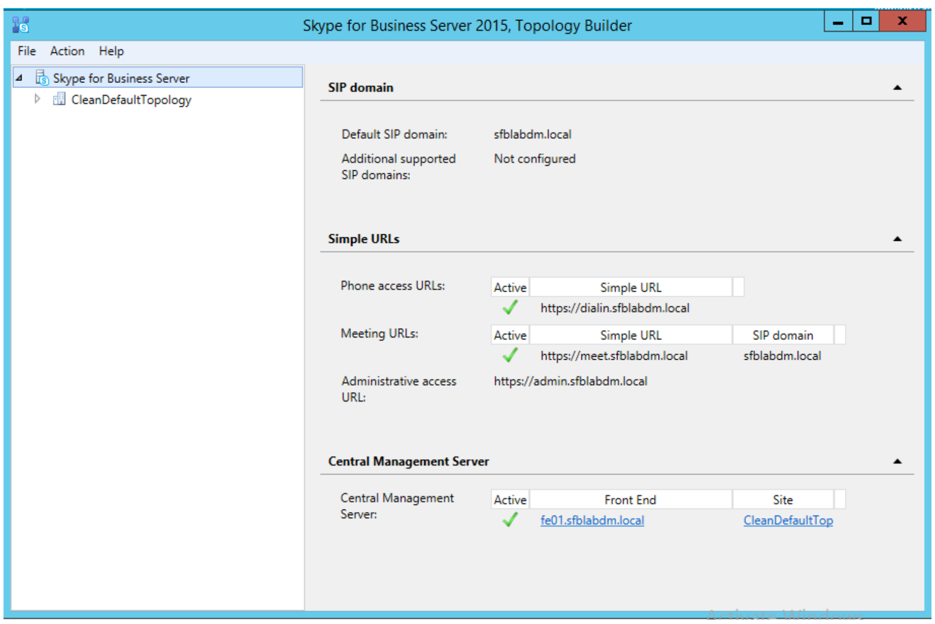


6. Next you will be prompted to save the topology which you have imported. You should revision the name or number of the topology according to the standards used within the enterprise. Click the **Save** button

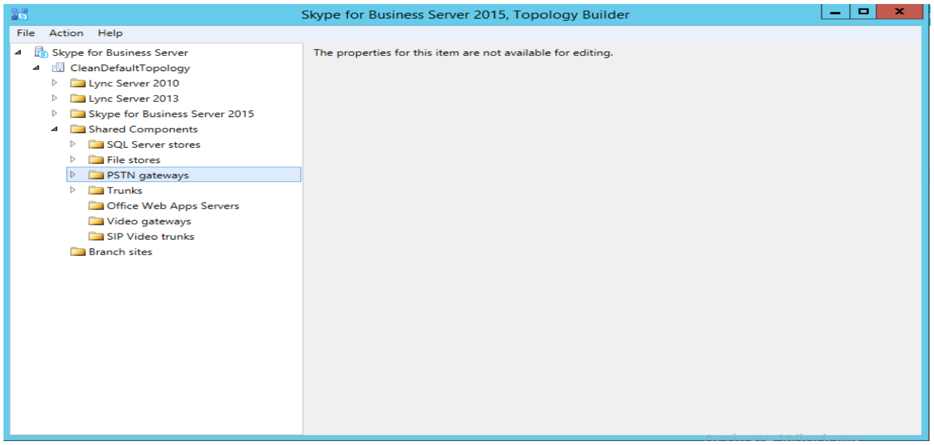
   Note: This keeps track of topology changes and, if desired, will allow you to fall back from any changes you make during this installation



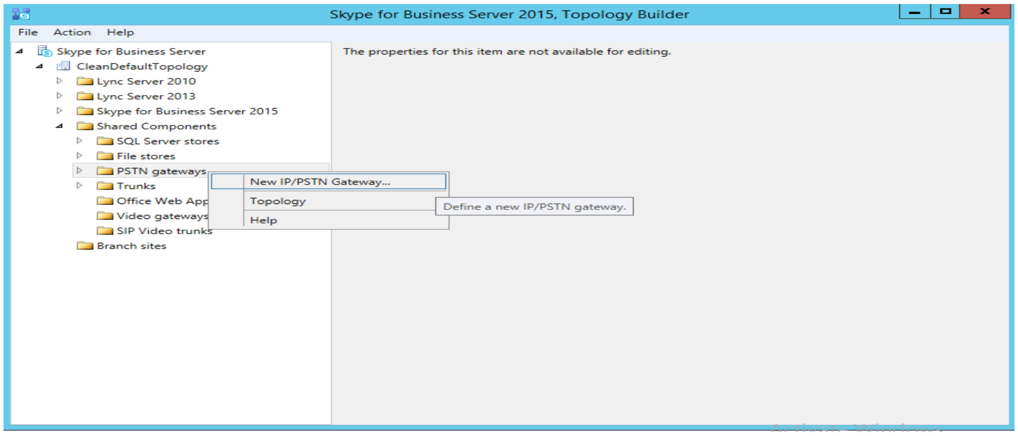
7. You will now see the topology builder screen with the enterprise's topology imported.



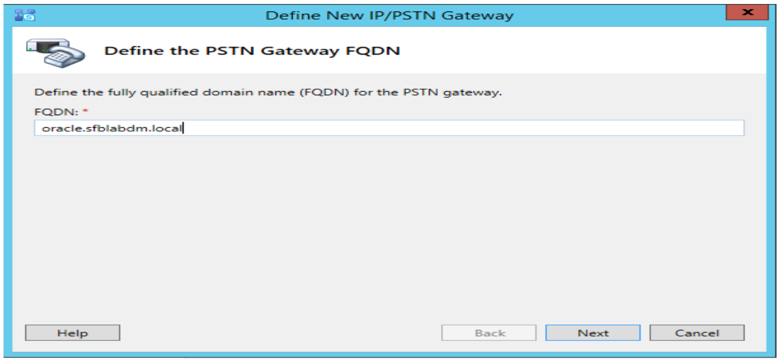
8. In the upper left hand corner, expand the site in which the PSTN gateway will be added. In our case, the site is **Bedford**. Then click on the **PSTN Gateways**



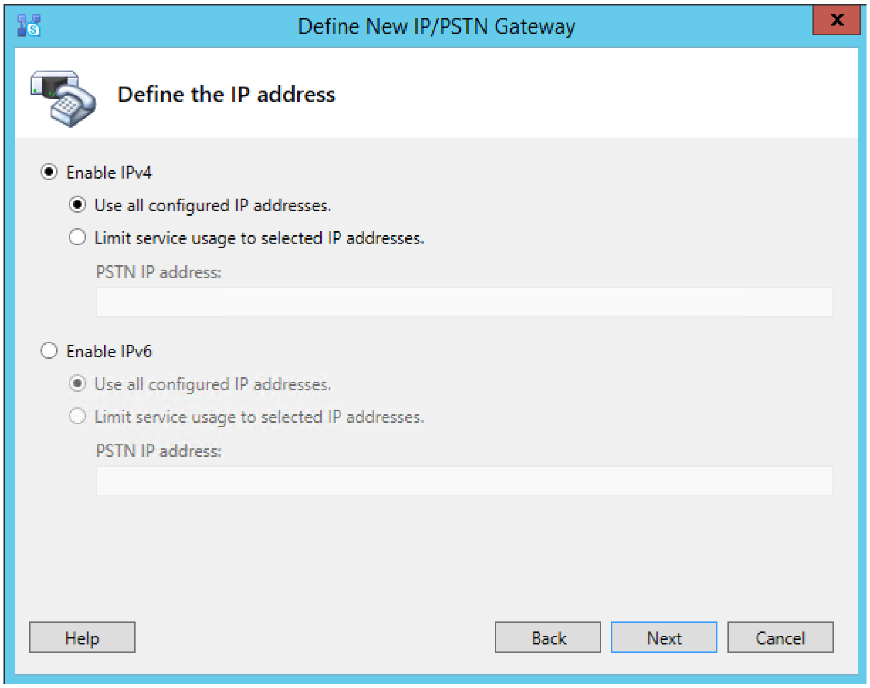
9. Right click on **PSTN gateways** and select **New IP/PSTN Gateway**.



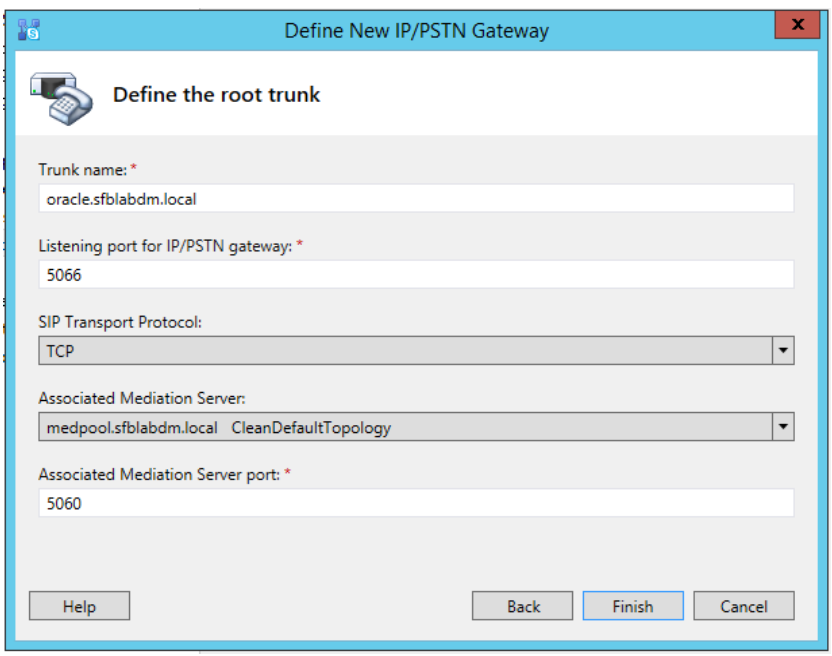
10. In the **Define New IP/PSTN Gateway** window, enter the ip address of the SIP interface of the SBC in the **FQDN** text box and click **Next**.



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



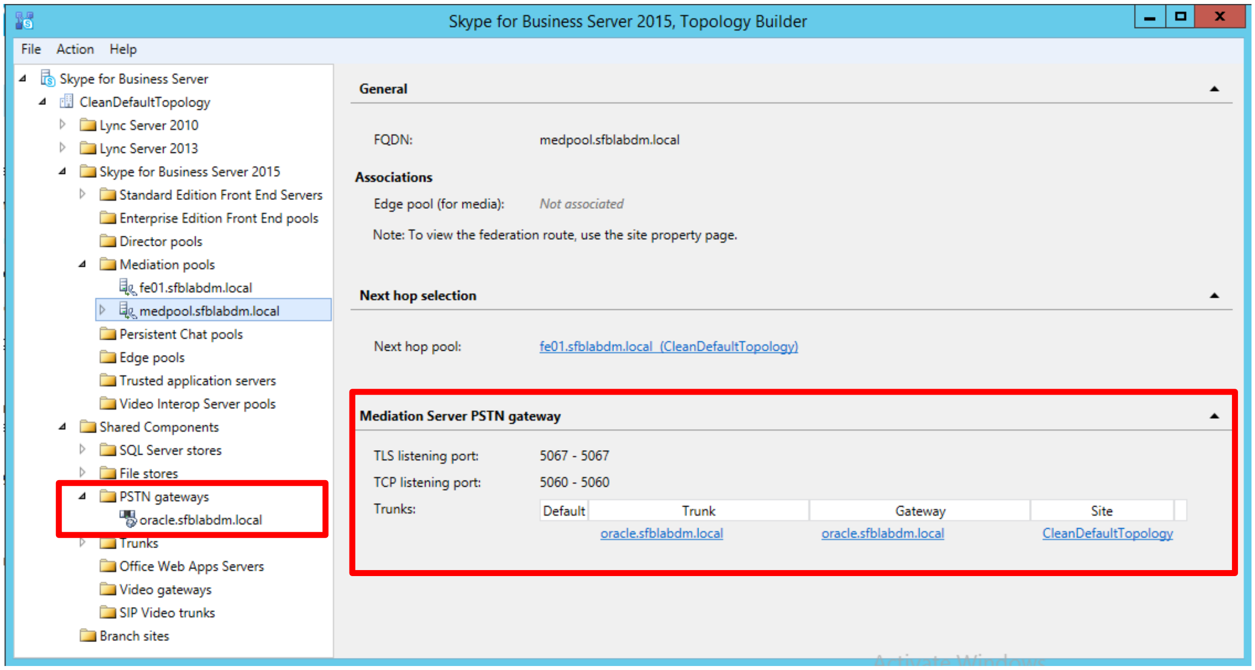
12. In the next section, enter the ip address of the SBC's sip interface under **Trunk name**. Configure the **Listening port for IP/PSTN gateway** as 5068, TCP as the **SIP Transport Protocol** and click **Finish**.



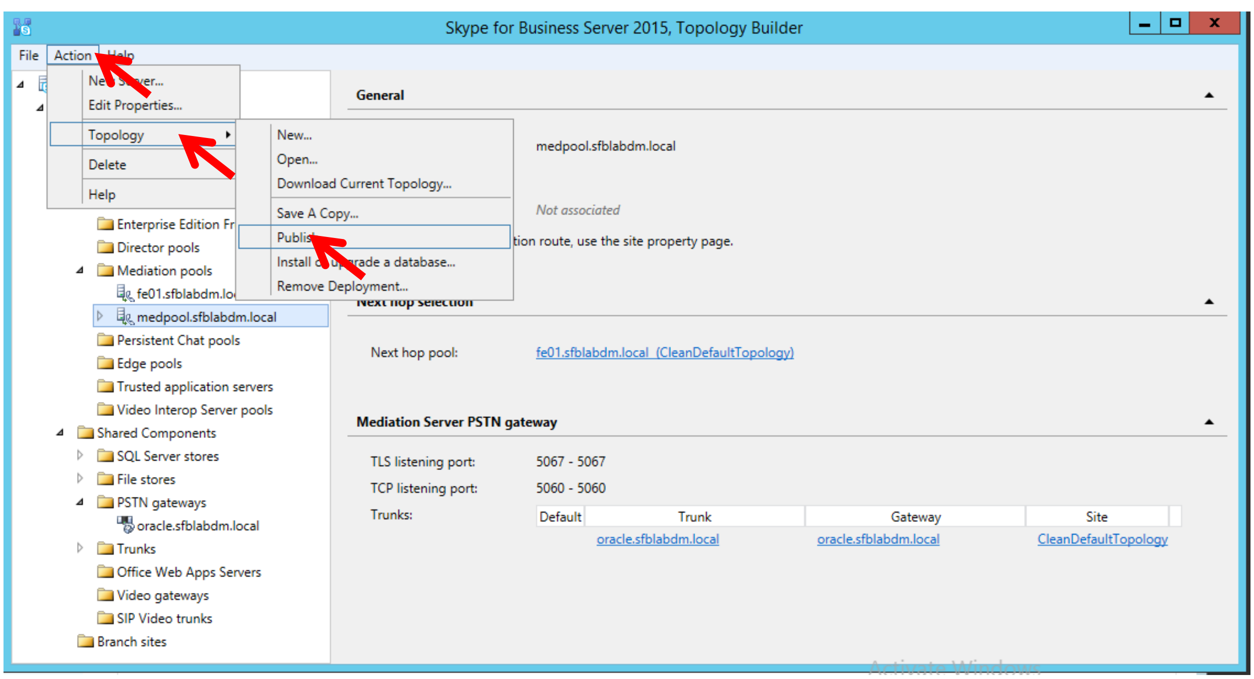
13. The PSTN gateway for the Lync server has been added. It will be listed under **PSTN gateways**.

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

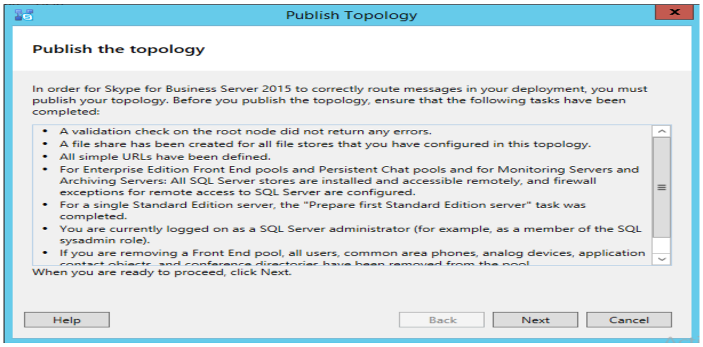
    You will see that the PSTN gateway is associated with the Mediation server.



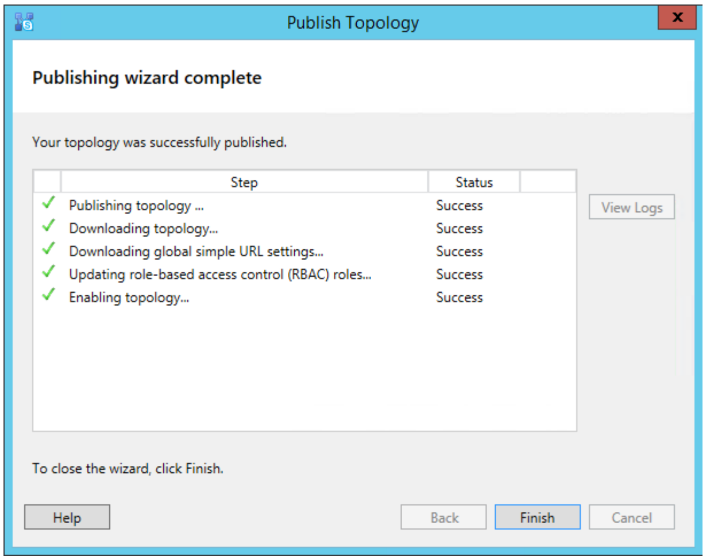
14. In the upper right hand corner of your screen under **Actions** select **Topology** then select **Publish**.



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



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



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

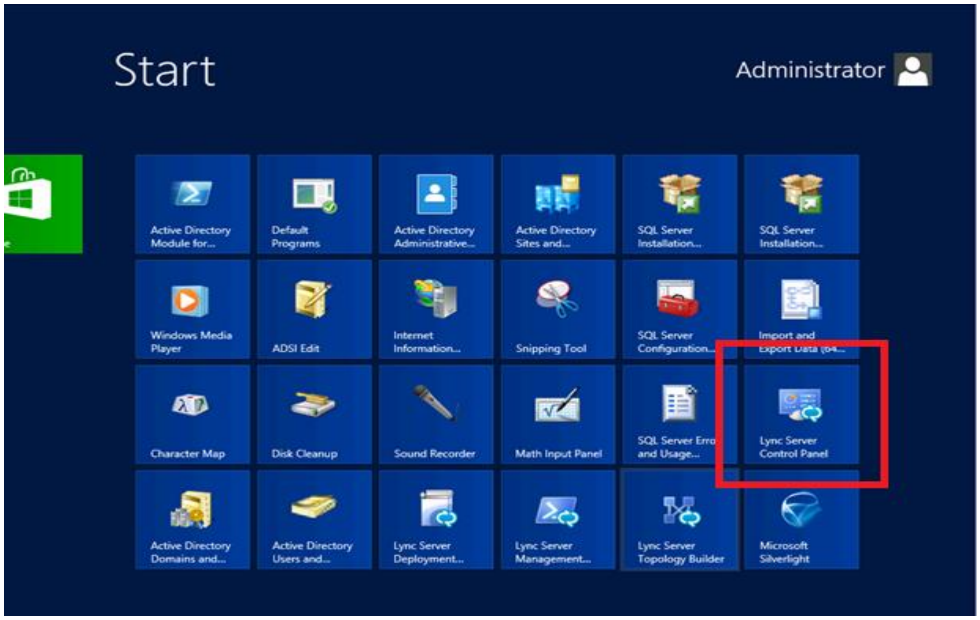
In order for the Lync Server Enterprise Voice clients to utilize the SIP trunking infrastructure that has been put in place, a route will need to be created to allow direction to this egress. Routes specify how Lync Server handles calls placed by enterprise voice users. When a user places a call, the 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:

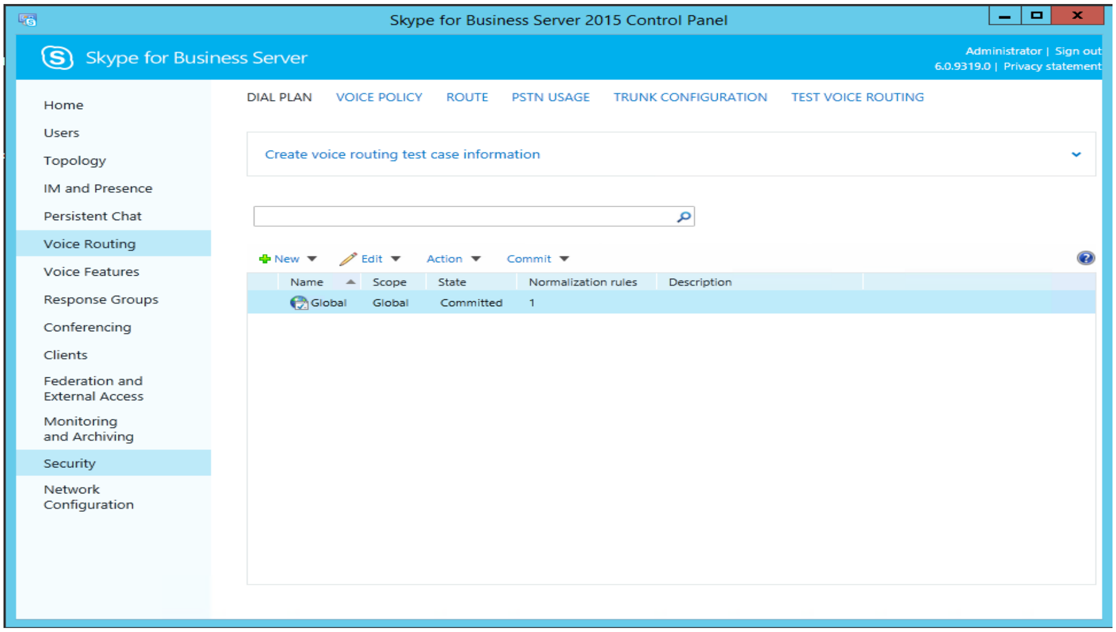
1. From the **Start** bar, select **Lync Server Control Panel**.



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**.



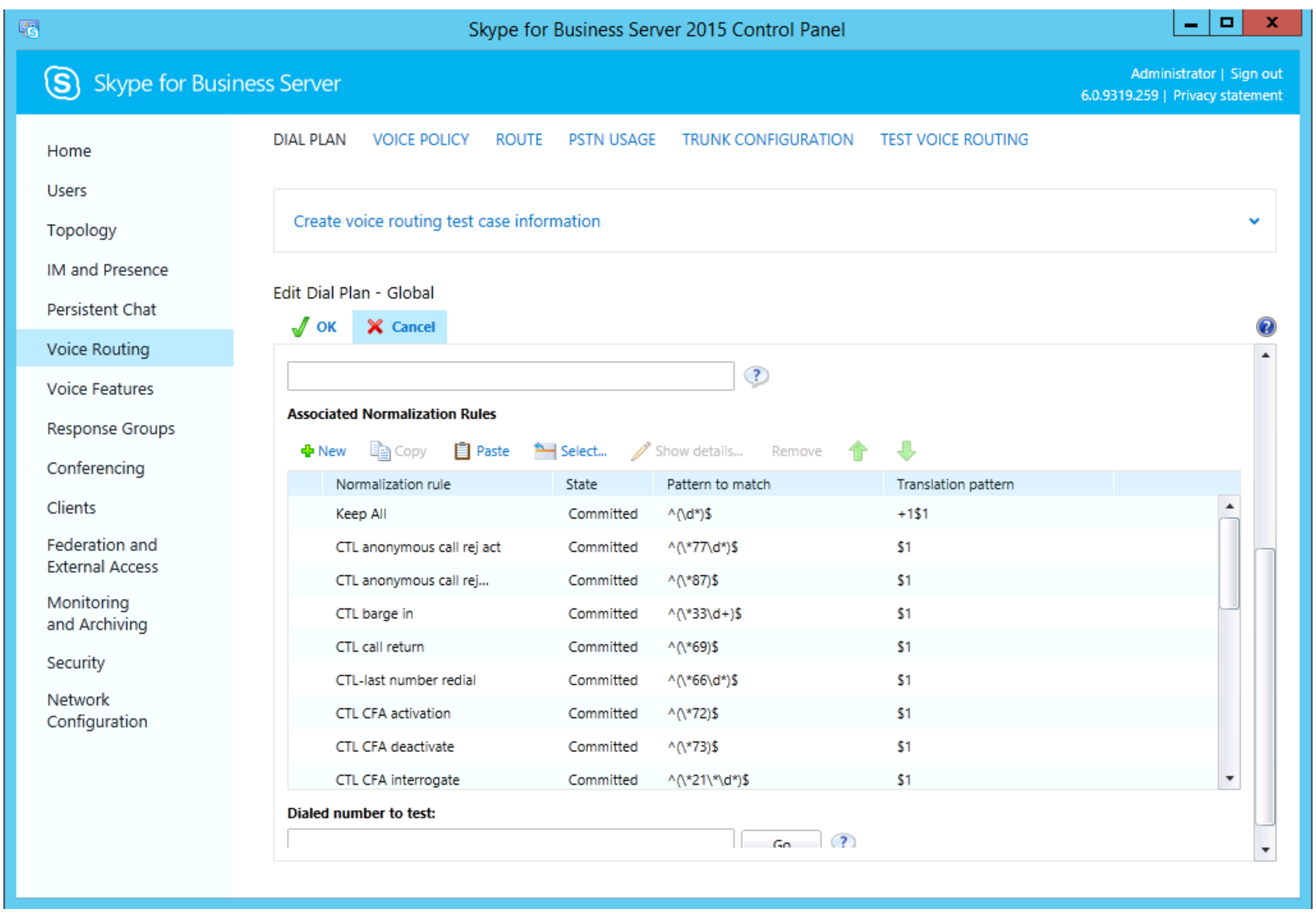
4. Next you build a Dial Plan and a translation rule for the phone numbers you want this route to handle.

US Dial-plan

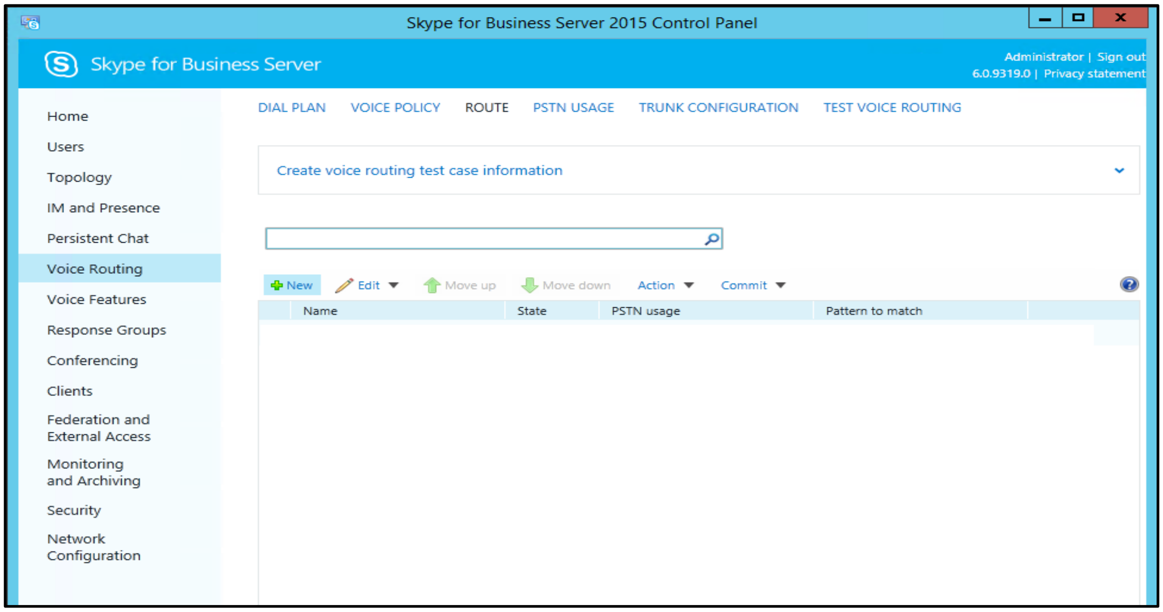
Match this pattern: ^(\d*)$

Translation rule: +1$1

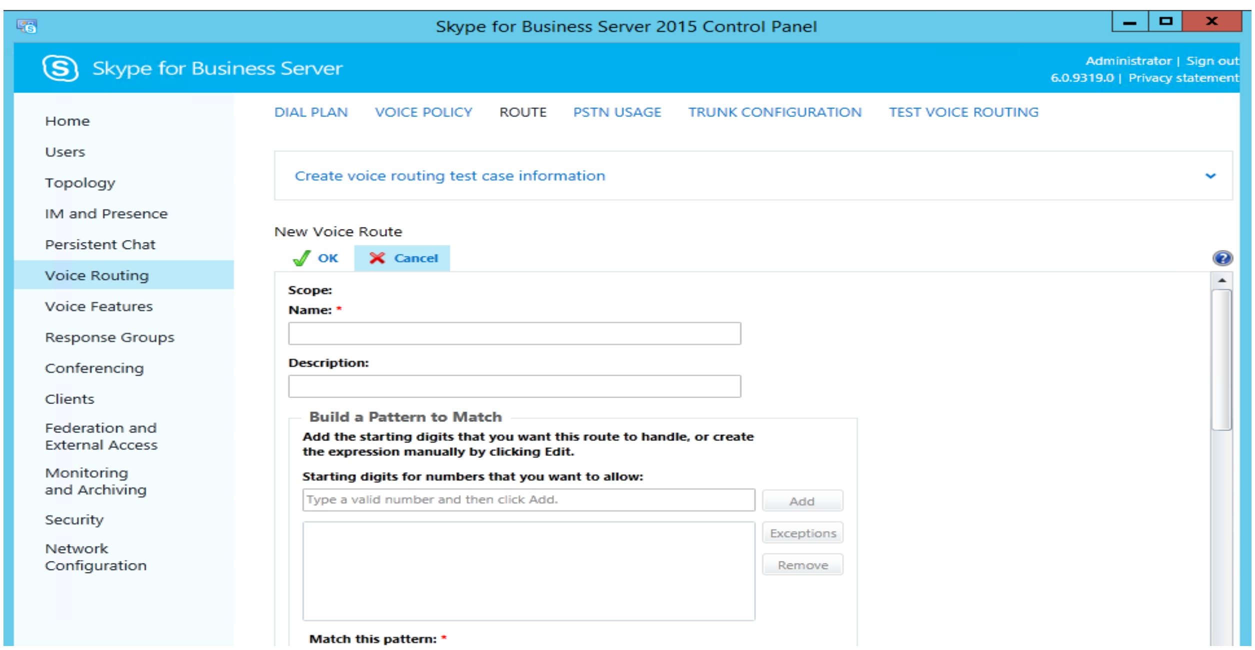
Additional normalization rules will need to be created as shown below to allow dialing the feature codes.



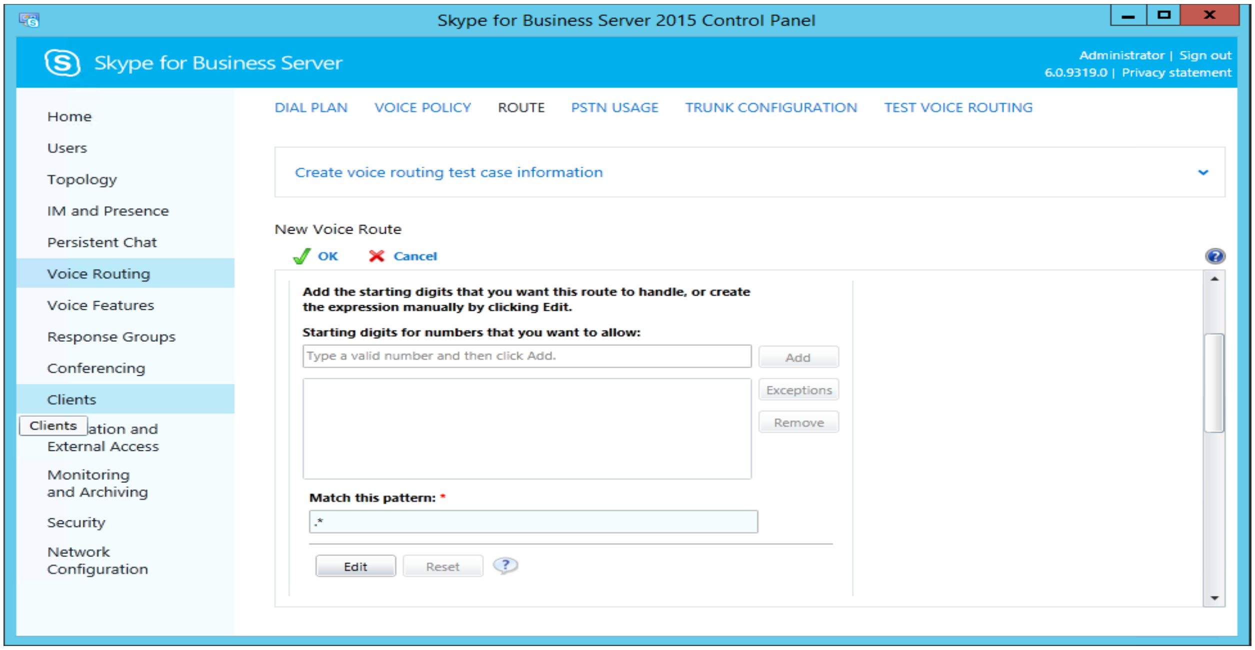
5. On the top row of the tabs, select **Route**. On the content area toolbar, click **+New**.



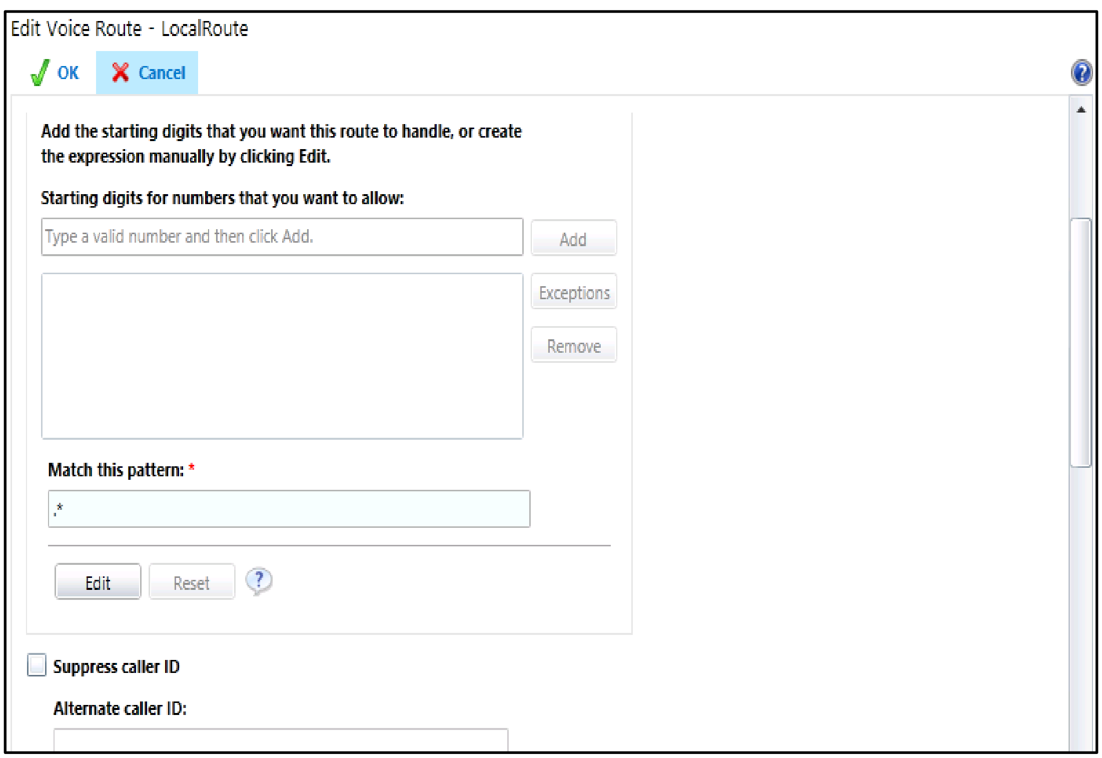
6. On the **New Voice Route** page, in the **Name** field, enter the name you have selected for the Route. In our example, it is US route.



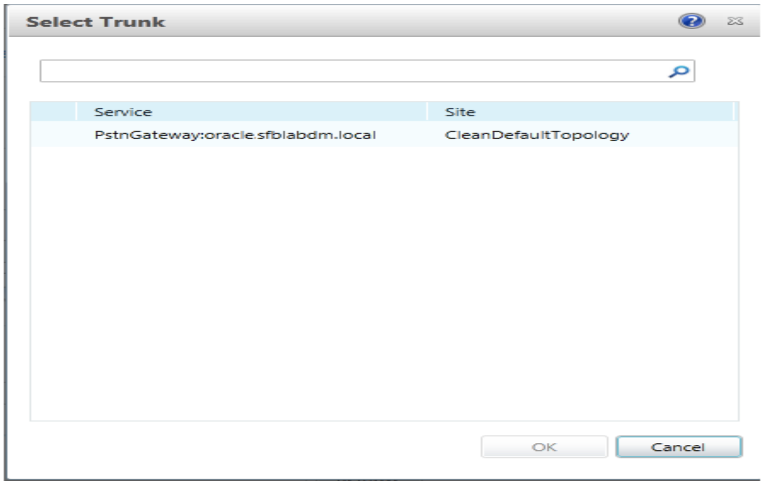
7. Next you build a Pattern Match for the phone numbers you want this route to handle. Click **Edit**.



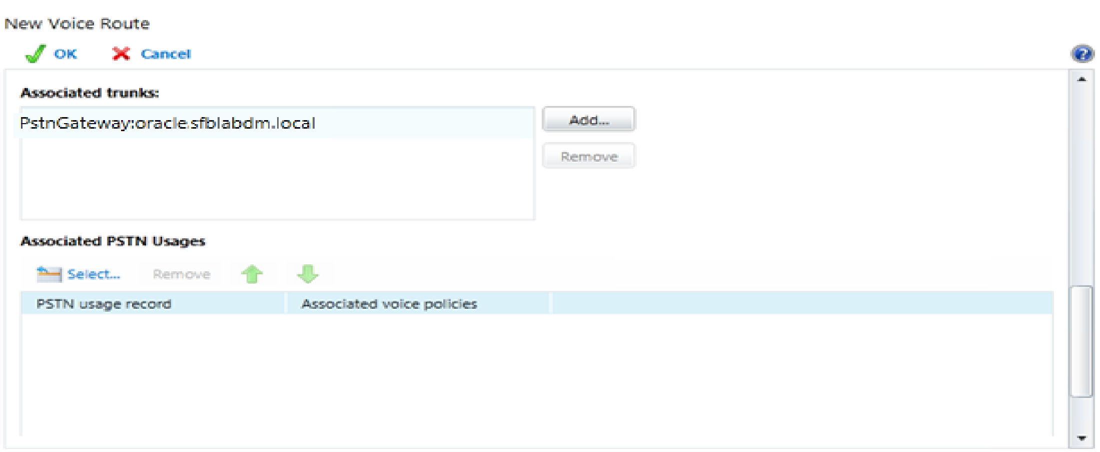
8. Next you want to associate the Voice Route with the **Trunk** you have just created. Scroll down to **Associated Trunks**, click on the **Add** button.



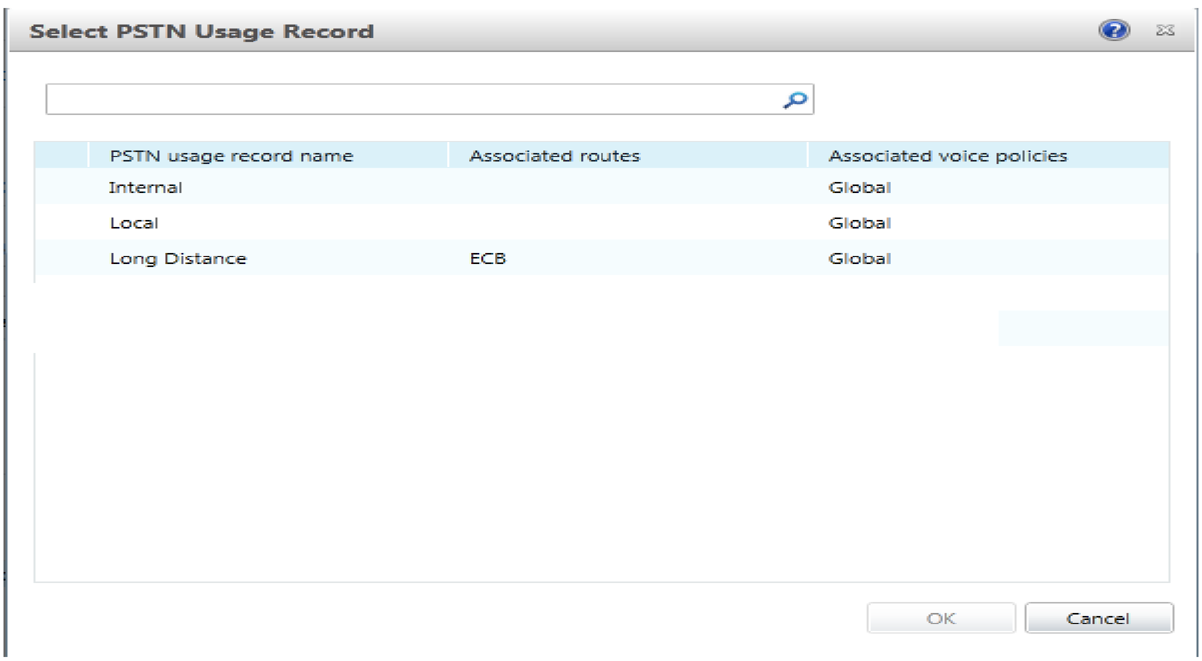
9. You will now be at a window showing available Trunks to associate your Voice Route. Click on the PSTN gateway that you just created and then click the **OK** button.



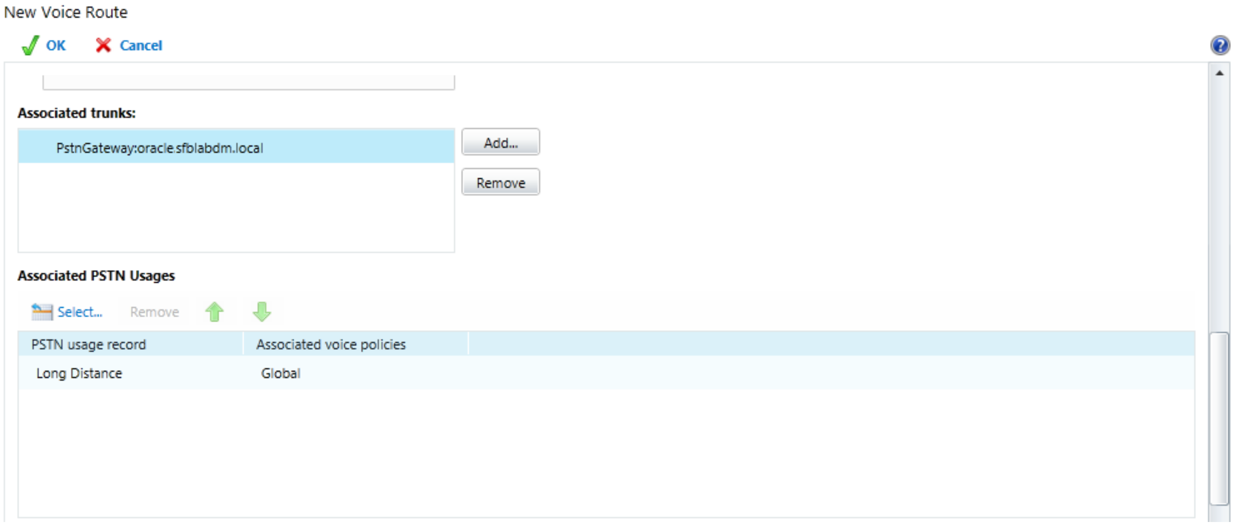
10. You can now see that you have associated your trunk with the route you created. An appropriate PSTN usage record will need to be assigned as well. In our example, we use one that was already created in the enterprise. Click on the **Select** button under **Associated PSTN Usages**.



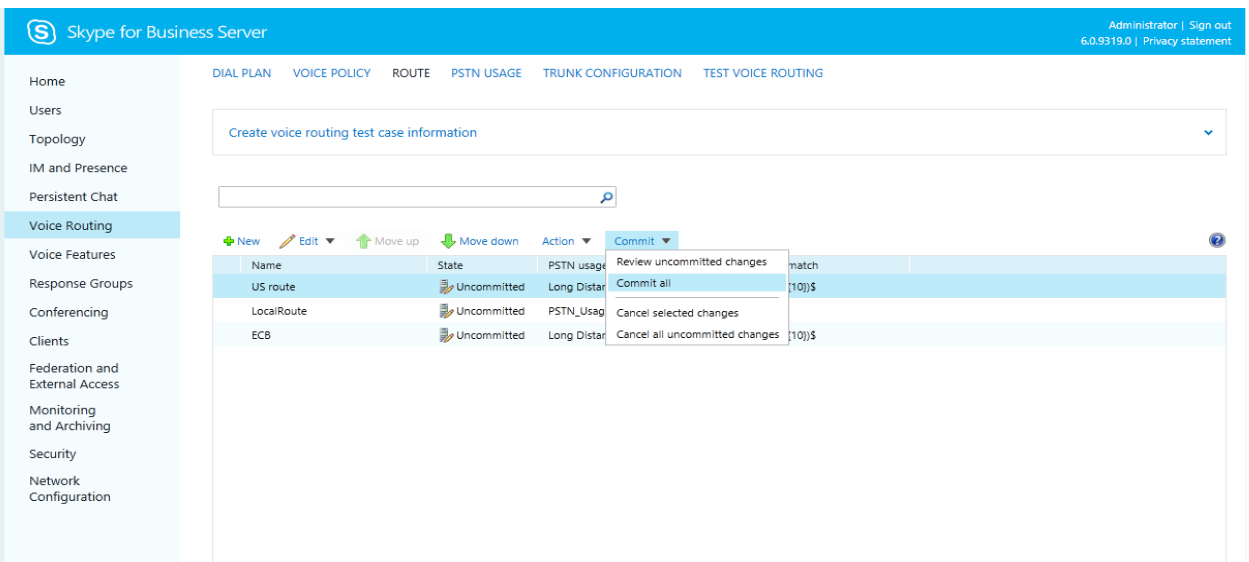
11. In the **Select PSTN Usage Record** window displayed, select the appropriate PSTN Usage Record and click **OK**.



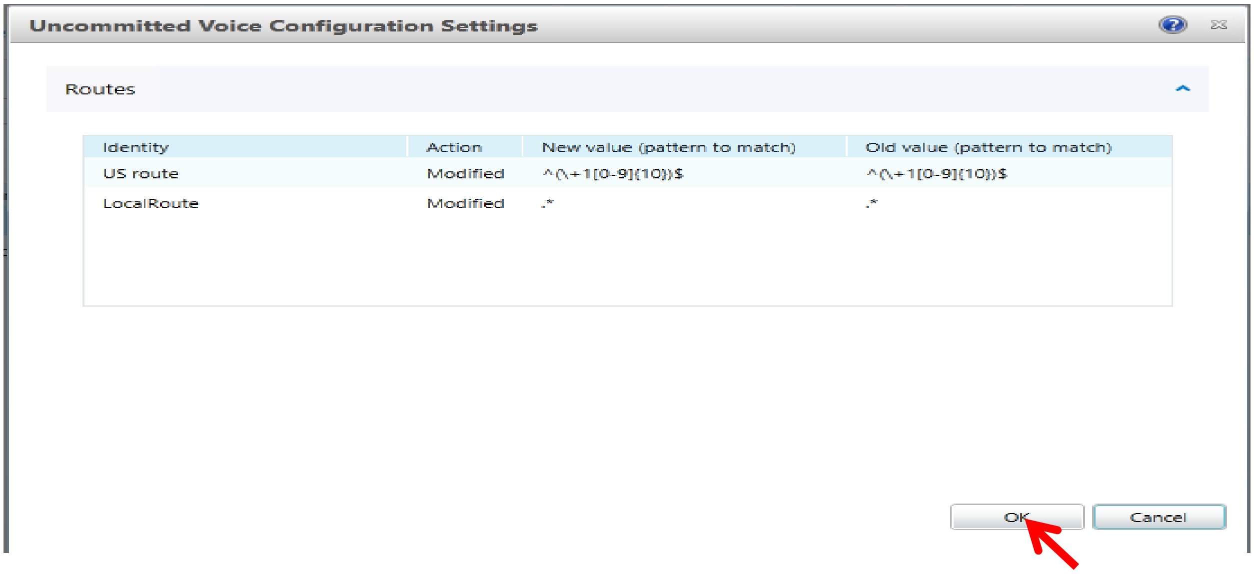
12. You will now see the Associated PSTN Usages which you have added. Click the **OK** button at the top of the **New Voice Route** screen.



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



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



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 http://docs.oracle.com/cd/E61547_01/index.html 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

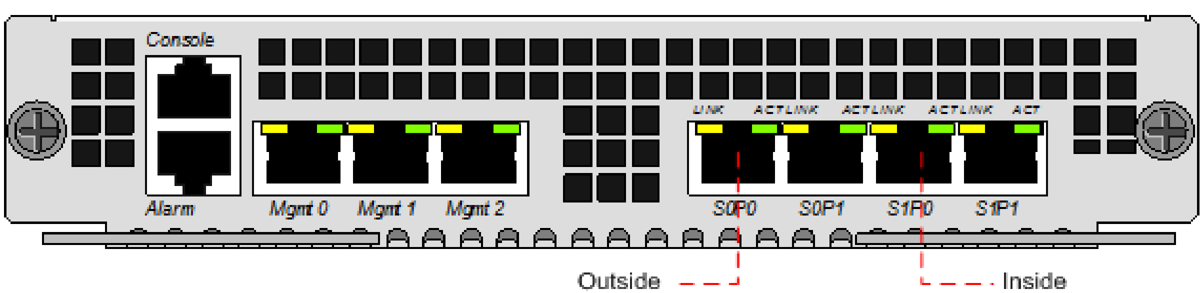
- Configuration of Network management including SNMP and RADIUS; and

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.

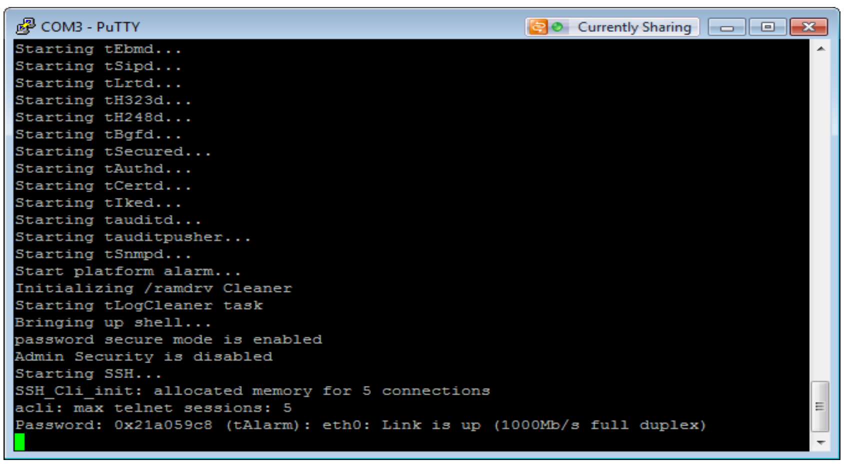
All commands are in bold, such as **`configure terminal`**; parameters in bold red such as oraclesbc1 are parameters which are specific to an individual deployment. **Note:** The ACLI is case sensitive.

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.



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                : 255.255.255.224
```

```
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 parameters will not go into effect until reboot.
Also, be aware that some boot parameters may also be changed through
PHY and Network Interface 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 mis-match 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-policy
        from-address                        *
        to-address                          *
        source-realm                        core
        policy-attribute
                next-hop                            192.168.1.220
                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
                next-hop                            192.168.1.220
                realm                               ctl-trunk
local-policy
        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                          sip
                tos-value                               0x18
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
        icmp-address                            192.168.1.220
network-interface
        name                                    s1p0
        hostname
        ip-address                              192.168.4.135
        netmask                                 255.255.255.0
        gateway                                 192.168.4.1
        dns-ip-primary                          192.168.4.150
        dns-domain                              partnersfb.com
        hip-ip-list                             192.168.4.135
        icmp-address                            192.168.4.135
phy-interface
        name                                    s0p0
        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                                192.168.1.220
        ip-address                              192.168.1.220
        port                                    5100
        realm-id                                ctl-trunk
        out-translationid                       stripplus1
        out-manipulationid                      Manipto7016
session-agent
        hostname                                medpool.partnersfb.com
        port                                    5068
        transport-method                        StaticTCP
        realm-id                                core
        ping-method                             OPTIONS
        ping-interval                           30
        out-translationid                       addplus1
        in-manipulationid                       ForEarlyMedia
        out-manipulationid                      outManipToSFB
        refer-call-transfer                     enabled
        auth-attributes
                auth-realm                              voip.centurylink.com
                username                                257570-4695550016
                password                                ********
                in-dialog-methods                       INVITE
session-translation
        id                                      addplus1
        rules-calling                           addplus1
        rules-called                            addplus1
session-translation
        id                                      stripplus1
        rules-calling                           stripplus1
        rules-called                            stripplus1
sip-config
        home-realm-id                           core
        registrar-domain                        *
        registrar-host                          *
        registrar-port                          5060
        options                                 inmanip-before-validate
                                                max-udp-length=0
        sip-message-len                         6000
        refer-src-routing                       enabled
sip-feature
        name                                    100rel-interworking
        realm                                   ctl-trunk
        require-mode-inbound                    Pass
        require-mode-outbound                   Pass
sip-interface
        realm-id                                core
        sip-port
                address                                 192.168.4.135
                transport-protocol                      TCP
                allow-anonymous                         agents-only
        registration-caching                    enabled
        options                                 100rel-interworking
        spl-options                             playback-on-refer="transferrbt"
sip-interface
        realm-id                                ctl-trunk
        sip-port
```

```
                address                                 192.168.1.220
                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
                name                                    ChechForPrivacy
                header-name                             Privacy
                action                                  manipulate
                msg-type                                request
                methods                                 INVITE
        header-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                                 INVITE
        header-rule
                name                                    addrequireinINVITE
                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
        name                                    Manipto7016
        header-rule
                name                                    modURI
                header-name                             request-uri
                action                                  manipulate
                element-rule
                        name                                    mod2
                        type                                    uri-host
                        action                                  replace
                        match-val-type                          fqdn
                        new-value                               $REMOTE_IP+:+$REMOTE_PORT
        header-rule
                name                                    DeletePAI
                header-name                             P-Asserted-Identity
                action                                  delete
                methods                                 INVITE
        header-rule
                name                                    ReplacePAI
                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
                methods                                 INVITE
                element-rule
                        name                                    From_add_voip_ctl_com
                        type                                    uri-host
                        action                                  replace
                        new-value                               voip.centurylink.com
        header-rule
                name                                    registercontactlocalipport
                header-name                             Contact
                action                                  manipulate
                msg-type                                request
                methods                                 REGISTER
                element-rule
                        name                                    registercontactlocalip
                        type                                    uri-host
                        action                                  replace
                        match-val-type                          ip
                        new-value                               $LOCAL_IP
                element-rule
```

```
                name                                        registercontactlocalport
                type                                        uri-port
                action                                      replace
                new-value                                   $LOCAL_PORT
        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
            header-name                                 Route
            action                                      manipulate
            element-rule
                name                                        updateroute2
                type                                        uri-user
                action                                      replace
                comparison-type                             pattern-rule
                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
                action                                      replace
                new-value                                   $LOCAL_IP
            element-rule
                name                                        modanonfrom
                type                                        uri-user
                action                                      replace
                comparison-type                             pattern-rule
                match-value                                 \+1anonymous
                new-value                                   anonymous
        header-rule
            name                                        To
            header-name                                 To
            action                                      manipulate
            msg-type                                    request
```

```
                element-rule
                        name                                        To
                        type                                        uri-host
                        action                                      replace
                        new-value                                   $REMOTE_IP
        header-rule
                name                                        fixcontactuser
                header-name                                 Contact
                action                                      manipulate
                comparison-type                             pattern-rule
                element-rule
                        name                                        user
                        type                                        uri-user
                        action                                      replace
                        comparison-type                             pattern-rule
                        match-value                                 (.*)
                        new-value                                   "+1"+$ORIGINAL
sip-monitoring
        match-any-filter                            enabled
steering-pool
        ip-address                                  192.168.1.220
        start-port                                  20000
        end-port                                    40000
        realm-id                                    ctl-trunk
steering-pool
        ip-address                                  192.168.4.135
        start-port                                  20000
        end-port                                    40000
        realm-id                                    core
surrogate-agent
        register-host                               voip.centurylink.com
        register-user                               4695550016
        realm-id                                    core
        customer-host                               192.168.1.220
        customer-next-hop                           192.168.1.220
        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
                state                                       enabled
                monitor-collector
                        address                                     172.18.255.71
        default-gateway                             192.168.1.1
translation-rules
        id                                          addplus1
        type                                        add
        add-string                                  +1
translation-rules
        id                                          stripplus1
        type                                        delete
        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 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 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 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 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 ringing 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 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 2 receives ringback PBX User 1 receives caller ID PBX User 1 hear Call Waiting Tone PBX User Places PSTN User 1 on Hold PSTN 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

MOH	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

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 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	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 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 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 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 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 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 ringing 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 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 2 receives ringback PBX User 1 receives caller ID PBX User 1 hear Call Waiting Tone PBX User Places PSTN User 1 on Hold PSTN 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
MOH	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

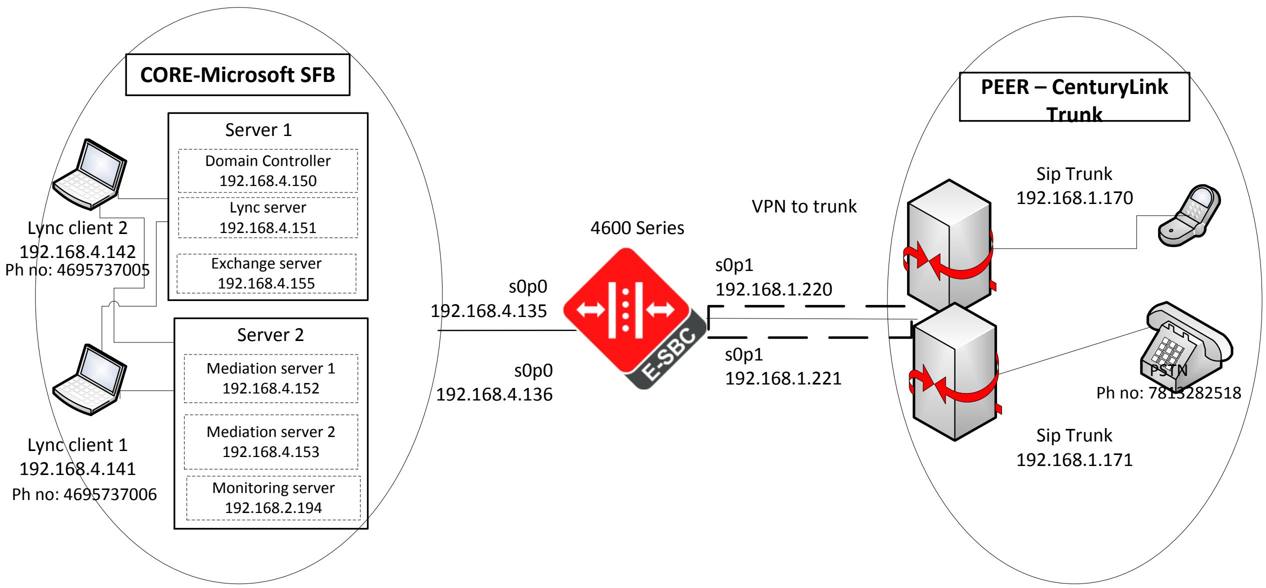
Simultaneous Ring	Provision Simultaneous ring in the BroadWorks portal. Place a call to the SIP trunk user. Verify that the numbers in the Simultaneous ring list are dialed at once.	Log into the Portal for PBX User 1 Set up Simultaneous Ring with PBX User 2 and PBX User 3 PSTN Calls PBX User 1 PSTN receives 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 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 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.

SBC configuration for Dual Trunk registration using PCMU towards trunk

```
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
                next-hop                                      192.168.1.170
                realm                                         ctl-trunk
local-policy
        from-address                                  *
        to-address                                    *
        source-realm                                  core2
        policy-attribute
                next-hop                                      192.168.1.171
                realm                                         ctl-trunk2
local-policy
        from-address                                  *
        to-address                                    SFBMed1.partnersfb.com
        source-realm                                  core2
        policy-attribute
                next-hop                                      SFBMed1.partnersfb.com
                realm                                         core2
                action                                        replace-uri
local-policy
        from-address                                  *
        to-address                                    SFBMed2.partnersfb.com
        source-realm                                  core2
        policy-attribute
                next-hop                                      SFBMed2.partnersfb.com
                realm                                         core2
                action                                        replace-uri
local-policy
        from-address                                  4695737015
        to-address                                    *
        source-realm                                  core2
        policy-attribute
                next-hop                                      192.168.1.171
                realm                                         ctl-trunk2
local-policy
        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                                  sip
                tos-value                                       0x18

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.partnersfb.com
        ip-address                                      192.168.4.135
        netmask                                         255.255.255.0
        gateway                                         192.168.4.1
        dns-ip-primary                                  192.168.4.150
        dns-domain                                      partnersfb.com
        hip-ip-list                                     192.168.4.135
                                                        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                                            s0p0
        operation-type                                  Media
phy-interface
        name                                            s1p0
        operation-type                                  Media
        slot                                            1
phy-interface
        name                                            s1p1
        operation-type                                  Media
        port                                            1
        slot                                            1
playback-config
        name                                            transferrbt
        entry
                encoding                                        PCMU
                filename                                        US_ringbackPCMU.raw

realm-config
        identifier                                      ctl-trunk2
```

```
        network-interfaces                      s0p0:0
        mm-in-realm                             enabled
        qos-enable                              enabled
        media-policy                            voip-default
realm-config
        identifier                              core
        network-interfaces                      s1p0:0
        mm-in-realm                             enabled
        qos-enable                              enabled
        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
        hostname                                192.168.1.171
        ip-address                              192.168.1.171
        port                                    5100
        realm-id                                ctl-trunk2
        out-translationid                       stripplus1
        out-manipulationid                      Manipto7015
session-agent
        hostname                                192.168.1.170
        ip-address                              192.168.1.170
        port                                    5100
        realm-id                                ctl-trunk
        out-translationid                       stripplus1
        out-manipulationid                      Manipto7016
session-agent
        hostname                                SFBMed1.partnersfb.com
        ip-address                              192.168.4.152
        port                                    5069
        transport-method                        StaticTCP
        realm-id                                core2
        ping-method                             OPTIONS
        ping-interval                           30
        out-translationid                       addplus1
        in-manipulationid                       ForEarlyMedia
        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
        ip-address                              192.168.4.153
```

```
        port                                        5069
        transport-method                            StaticTCP
        realm-id                                    core2
        ping-method                                 OPTIONS
        ping-interval                               30
        out-translationid                           addplus1
        in-manipulationid                           ForEarlyMedia
        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                                    medpool.partnersfb.com
        port                                        5068
        transport-method                            StaticTCP
        realm-id                                    core
        ping-method                                 OPTIONS
        ping-interval                               30
        out-translationid                           addplus1
        in-manipulationid                           ForEarlyMedia
        out-manipulationid                          outManipToSFB
        refer-call-transfer                         enabled
        auth-attributes
                auth-realm                                  voip.centurylink.com
                username                                    257570-4695550016
                password                                    ********
                in-dialog-methods                           INVITE
session-group
        group-name                                  SFBmed
        strategy                                    RoundRobin
        dest                                        SFBMed1.partnersfb.com
                                                    SFBMed2.partnersfb.com
session-group
        group-name                                  ctl
        dest                                        192.168.1.171
                                                    192.168.1.170
session-translation
        id                                          addplus1
        rules-calling                               addplus1
        rules-called                                addplus1
session-translation
        id                                          stripplus1
        rules-calling                               stripplus1
        rules-called                                stripplus1
sip-config
        home-realm-id                               core
        registrar-domain                            *
        registrar-host                              *
        registrar-port                              5060
        options                                     inmanip-before-validate
                                                    max-udp-length=0
        sip-message-len                             6000
        refer-src-routing                           enabled
```

```
sip-feature
        name                                        100rel-interworking
        realm                                       ctl-trunk
        require-mode-inbound                        Pass
        require-mode-outbound                       Pass

sip-interface
        realm-id                                    ctl-trunk2
        sip-port
                address                                     192.168.1.221
                allow-anonymous                             agents-only
sip-interface
        realm-id                                    core
        sip-port
                address                                     192.168.4.135
                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
                address                                     192.168.4.136
                transport-protocol                          TCP
        registration-caching                        enabled
        options                                     100rel-interworking
        spl-options                                 playback-on-refer="transferrbt"
sip-interface
        realm-id                                    ctl-trunk
        sip-port
                address                                     192.168.1.220
                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
                name                                        ChechForPrivacy
                header-name                                 Privacy
                action                                      manipulate
                msg-type                                    request
```

```
                methods                                 INVITE
        header-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                                 INVITE
        header-rule
                name                                    addrequireinINVITE
                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
        name                                    Manipto7015
        header-rule
                name                                    modURI
                header-name                             request-uri
                action                                  manipulate
                element-rule
                        name                                    mod2
                        type                                    uri-host
                        action                                  replace
                        match-val-type                          fqdn
                        new-value                               $REMOTE_IP+:+$REMOTE_PORT
        header-rule
                name                                    DeletePAI
                header-name                             P-Asserted-Identity
                action                                  delete
                methods                                 INVITE
```

```
    header-rule
            name                                    ReplacePAI
            header-name                             P-Asserted-Identity
            action                                  add
            comparison-type                         pattern-rule
            methods                                 INVITE
            new-value                               sip:4695737015@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
                    type                                    uri-host
                    action                                  replace
                    new-value                               voip.centurylink.com
    header-rule
            name                                    registercontactlocalipport
            header-name                             Contact
            action                                  manipulate
            msg-type                                request
            methods                                 REGISTER
            element-rule
                    name                                    registercontactlocalip
                    type                                    uri-host
                    action                                  replace
                    match-val-type                          ip
                    new-value                               $LOCAL_IP
            element-rule
                    name                                    registercontactlocalport
                    type                                    uri-port
                    action                                  replace
                    new-value                               $LOCAL_PORT
    header-rule
            name                                    fixContact1
            header-name                             Contact
            action                                  manipulate
            element-rule
                    name                                    updatecon
                    type                                    uri-user
                    action                                  replace
                    comparison-type                         pattern-rule
                    match-value                             (.*)
                    new-value                               $ORIGINAL-^"+1"
    header-rule
            name                                    fixRoute1
            header-name                             Route
            action                                  manipulate
            element-rule
                    name                                    updateroute
                    type                                    uri-user
                    action                                  replace
                    comparison-type                         pattern-rule
                    match-value                             (.*)
                    new-value                               $ORIGINAL-^"+1"
```

```
        header-rule
                name                                    Privacy
                header-name                             From
                action                                  sip-manip
                new-value                               Check_privacy_header
sip-manipulation
        name                                    Manipto7016
        header-rule
                name                                    modURI
                header-name                             request-uri
                action                                  manipulate
                element-rule
                        name                                    mod2
                        type                                    uri-host
                        action                                  replace
                        match-val-type                          fqdn
                        new-value                               $REMOTE_IP+:+$REMOTE_PORT
        header-rule
                name                                    DeletePAI
                header-name                             P-Asserted-Identity
                action                                  delete
                methods                                 INVITE
        header-rule
                name                                    ReplacePAI
                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
                methods                                 INVITE
                element-rule
                        name                                    From_add_voip_ctl_com
                        type                                    uri-host
                        action                                  replace
                        new-value                               voip.centurylink.com
        header-rule
                name                                    registercontactlocalipport
                header-name                             Contact
                action                                  manipulate
                msg-type                                request
                methods                                 REGISTER
                element-rule
                        name                                    registercontactlocalip
                        type                                    uri-host
                        action                                  replace
                        match-val-type                          ip
                        new-value                               $LOCAL_IP
                element-rule
                        name                                    registercontactlocalport
                        type                                    uri-port
                        action                                  replace
                        new-value                               $LOCAL_PORT
```

```
        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
                header-name                                 Route
                action                                      manipulate
                element-rule
                        name                                        updateroute2
                        type                                        uri-user
                        action                                      replace
                        comparison-type                             pattern-rule
                        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
                        action                                      replace
                        new-value                                   $LOCAL_IP
                element-rule
                        name                                        modanonfrom
                        type                                        uri-user
                        action                                      replace
                        comparison-type                             pattern-rule
                        match-value                                 \+1anonymous
                        new-value                                   anonymous
        header-rule
                name                                        To
                header-name                                 To
                action                                      manipulate
                msg-type                                    request
                element-rule
                        name                                        To
                        type                                        uri-host
                        action                                      replace
```

```
                        new-value                                      $REMOTE_IP
                header-rule
                        name                                    fixcontactuser
                        header-name                             Contact
                        action                                  manipulate
                        comparison-type                         pattern-rule
                        element-rule
                                name                                    user
                                type                                    uri-user
                                action                                  replace
                                comparison-type                         pattern-rule
                                match-value                             (.*)
                                new-value                               "+1"+$ORIGINAL


sip-monitoring
        match-any-filter                        enabled
steering-pool
        ip-address                              192.168.1.220
        start-port                              20000
        end-port                                40000
        realm-id                                ctl-trunk
steering-pool
        ip-address                              192.168.1.221
        start-port                              20000
        end-port                                40000
        realm-id                                ctl-trunk2
steering-pool
        ip-address                              192.168.3.220
        start-port                              20000
        end-port                                40000
        realm-id                                Nice-SIPREC
steering-pool
        ip-address                              192.168.4.135
        start-port                              20000
        end-port                                40000
        realm-id                                core
steering-pool
        ip-address                              192.168.4.136
        start-port                              20000
        end-port                                40000
        realm-id                                core2
surrogate-agent
        register-host                           voip.centurylink.com
        register-user                           4695737015
        realm-id                                core2
        customer-host                           192.168.1.171
        customer-next-hop                       192.168.1.171
        register-contact-host                   192.168.1.171
        register-contact-user                   4695737015
        password                                *************
        register-expires                        60
        auth-user                               257570-4695737015
surrogate-agent
        register-host                           voip.centurylink.com
        register-user                           4695550016
```

```
        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
                monitor-collector
                        address                                 172.18.255.71
        default-gateway                         192.168.1.1
translation-rules
        id                                      addplus1
        type                                    add
        add-string                              +1
translation-rules
        id                                      stripplus1
        type                                    delete
        delete-string                           +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	1. Each SBC is configured with a trunk to ITSP and associated authentication/digest and registration information. 2. Invoke a command on SBC to register the trunk with ITSP. 3. Verify that 200 OK is received from ITSP for both the trunks.	Passed

Inbound PSTN calls pick correct trunk to SBC	Verify that PSTN to PBX inbound calls arrive on both the trunks when multiple calls are made	1. Dial an inbound call to the PBX. 2. Verify ringing is heard by calling and called parties. 3. Verify the trace shows a valid ringing indication message 4. Take called party phone off-hook. 5. Verify that a media path is established in both directions. 6. Hang up calling party 7. Verify the IP/PBX 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.	Passed

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	1. Dial an outbound call from the PBX. 2. Verify ringing is heard by calling and called parties. 3. Verify the trace shows a valid ringing indication message 4. Take called party phone off-hook. 5. Verify that a media path established in both directions. 6. Hang up Calling Party 7. Verify the IP/PBX sends a Bye message. 8. Make a note of the Trunk on which the call was sent to ITSP. 9. Repeat the above steps 3 more times (total 4 calls). 10. Verify that calls from PBX are sent out on both the trunks to ITSP. 11. Verify each call has PAI sent per the trunk configuration 12. Document Test Results. 13. Save Trace.	Passed
Alien TN calls on 1st trunk	Verify calls are successful with Alien TNs on 1st trunk	1. After Alien TN is set up on a Trunk1 in CenturyLink Network 2. PBX User Places a Call to PSTN 3. PBX User receives ring back 4. PSTN receives ringing 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	Passed

Alien TN calls on 2nd trunk	Verify calls are successful with Alien TNs on 2nd trunk	1. After Alien TN is set up on a Trunk2 in CenturyLink Network 2. PBX User Places a Call to PSTN 3. PBX User receives ring back 4. PSTN receives ringing 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	Passed
Failover of 1st trunk WAN - PSTN-PBX	Ensure that calls are delivered from PSTN to PBX when the first trunk has failed on the WAN side	1. Down the WAN interface associated with Trunk 1. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 2 4. PBX user answers the 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	Passed

Failover of 1st trunk WAN - PBX-PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has failed on the WAN side	1. Down the WAN interface associated with Trunk 1. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that all 3 calls are delivered to PSTN utilizing Trunk 2 4. PSTN user answers the 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	Passed
Restore 1st trunk WAN: PSTN-PBX	Ensure that calls are delivered from PSTN to PBX when the first trunk has has been restored	1. WAN interface associated with Trunk 1 is brought back into service. 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 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	Passed

Restore 1st trunk WAN: PBX-PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has has been restored	1. WAN interface associated with Trunk 1 is brought back into service. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that at least one call is delivered to the PSTN via Trunk 1 4. PSTN user answers the 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	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	1. Down the WAN interface associated with Trunk 2. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 1 4. PBX user answers the 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	Passed

Failover of 2nd trunk WAN: PBX-PSTN	Ensure that calls are delivered from PBX-PSTN when the second trunk has failed on the WAN side	1. Down the WAN interface associated with Trunk 2. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 1 4. PSTN user answers the 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	Passed
Restore 2nd trunk WAN: PSTN-PBX	Ensure that calls are delivered from PSTN to PBX when the second trunk has has been restored	1. WAN interface associated with Trunk 2 is brought back into service. 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 2 4. PSTN user answers the 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	Passed

Restore 2nd trunk WAN: PBX-PSTN	Ensure that calls are delivered from PBX to PSTN when the second trunk has has been restored	1. WAN interface associated with Trunk 2 is brought back into service. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that at least one call is delivered to the PBX via Trunk 2 4. PBX user answers the 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	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	1. Down the LAN interface associated with Trunk 1. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that all 3 calls are delivered to PSTN utilizing Trunk 2 4. PSTN user answers the 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	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	1. Down the LAN interface associated with Trunk 1. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 2 4. PBX user answers the 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	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	1. LAN interface associated with Trunk 1 is brought back into service. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that at least one call is delivered to the PSTN via Trunk 2 4. PSTN user answers the 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	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	1. LAN interface associated with Trunk 1 is brought back into service. 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 2 4. PBX user answers the 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	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	1. Down the LAN interface associated with Trunk 2. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that all 3 calls are delivered to PSTN utilizing Trunk 1 4. PSTN user answers the 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	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	1. Down the LAN interface associated with Trunk 2. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 1 4. PBX user answers the 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	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	1. LAN interface associated with Trunk 2 is brought back into service. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that at least one call is delivered to the PSTN via Trunk 2 4. PSTN user answers the 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	Passed

Restore 2nd trunk LAN - PSTN to PBX	Ensure that calls are delivered from PSTN to PBX when the second trunk has been restored	1. LAN interface associated with Trunk 2 is brought back into service. 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 2 4. PBX user answers the 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	Passed
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	1. Each SBC is configured with a trunk to ITSP and associated authentiation/digest and registration information. 2. Invoke a command on SBC to register the trunk with ITSP. 3. Verify that 200 OK is received from ITSP for both the trunks.	Passed

Inbound PSTN calls pick correct trunk to SBC	Verify that PSTN to PBX inbound calls arrive on both the trunks when multiple calls are made	1. Dial an inbound call to the PBX. 2. Verify ringing is heard by calling and called parties. 3. Verify the trace shows a valid ringing indication message 4. Take called party phone off-hook. 5. Verify that a media path is established in both directions. 6. Hang up calling party 7. Verify the IP/PBX 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.	Passed

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	1. Dial an outbound call from the PBX. 2. Verify ringing is heard by calling and called parties. 3. Verify the trace shows a valid ringing indication message 4. Take called party phone off-hook. 5. Verify that a media path established in both directions. 6. Hang up Calling Party 7. Verify the IP/PBX sends a Bye message. 8. Make a note of the Trunk on which the call was sent to ITSP. 9. Repeat the above steps 3 more times (total 4 calls). 10. Verify that calls from PBX are sent out on both the trunks to ITSP. 11. Verify each call has PAI sent per the trunk configuration 12. Document Test Results. 13. Save Trace.	Passed
Alien TN calls on 1st trunk	Verify calls are successful with Alien TNs on 1st trunk	1. After Alien TN is set up on a Trunk1 in CenturyLink Network 2. PBX User Places a Call to PSTN 3. PBX User receives ring back 4. PSTN receives ringing 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	Passed

Alien TN calls on 2nd trunk	Verify calls are successful with Alien TNs on 2nd trunk	1. After Alien TN is set up on a Trunk2 in CenturyLink Network 2. PBX User Places a Call to PSTN 3. PBX User receives ring back 4. PSTN receives ringing 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	Passed
Failover of 1st trunk WAN - PSTN-PBX	Ensure that calls are delivered from PSTN to PBX when the first trunk has failed on the WAN side	1. Down the WAN interface associated with Trunk 1. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 2 4. PBX user answers the 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	Passed

Failover of 1st trunk WAN - PBX-PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has failed on the WAN side	1. Down the WAN interface associated with Trunk 1. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that all 3 calls are delivered to PSTN utilizing Trunk 2 4. PSTN user answers the 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	Passed
Restore 1st trunk WAN: PSTN-PBX	Ensure that calls are delivered from PSTN to PBX when the first trunk has has been restored	1. WAN interface associated with Trunk 1 is brought back into service. 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 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	Passed

Restore 1st trunk WAN: PBX-PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has has been restored	1. WAN interface associated with Trunk 1 is brought back into service. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that at least one call is delivered to the PSTN via Trunk 1 4. PSTN user answers the 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	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	1. Down the WAN interface associated with Trunk 2. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 1 4. PBX user answers the 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	Passed

Failover of 2nd trunk WAN: PBX-PSTN	Ensure that calls are delivered from PBX-PSTN when the second trunk has failed on the WAN side	1. Down the WAN interface associated with Trunk 2. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 1 4. PSTN user answers the 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	Passed
Restore 2nd trunk WAN: PSTN-PBX	Ensure that calls are delivered from PSTN to PBX when the second trunk has has been restored	1. WAN interface associated with Trunk 2 is brought back into service. 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 2 4. PSTN user answers the 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	Passed

Restore 2nd trunk WAN: PBX-PSTN	Ensure that calls are delivered from PBX to PSTN when the second trunk has has been restored	1. WAN interface associated with Trunk 2 is brought back into service. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that at least one call is delivered to the PBX via Trunk 2 4. PBX user answers the 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	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	1. Down the LAN interface associated with Trunk 1. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that all 3 calls are delivered to PSTN utilizing Trunk 2 4. PSTN user answers the 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	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	1. Down the LAN interface associated with Trunk 1. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 2 4. PBX user answers the 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	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	1. LAN interface associated with Trunk 1 is brought back into service. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that at least one call is delivered to the PSTN via Trunk 2 4. PSTN user answers the 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	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	1. LAN interface associated with Trunk 1 is brought back into service. 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 2 4. PBX user answers the 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	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	1. Down the LAN interface associated with Trunk 2. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that all 3 calls are delivered to PSTN utilizing Trunk 1 4. PSTN user answers the 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	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	1. Down the LAN interface associated with Trunk 2. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 1 4. PBX user answers the 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	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	1. LAN interface associated with Trunk 2 is brought back into service. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that at least one call is delivered to the PSTN via Trunk 2 4. PSTN user answers the 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	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	1. LAN interface associated with Trunk 2 is brought back into service. 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 2 4. PBX user answers the 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	Passed

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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 www.microsoft.com/downloads. 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 www.wireshark.org. 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 http://docs.oracle.com/cd/E56581_01/index.htm

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.

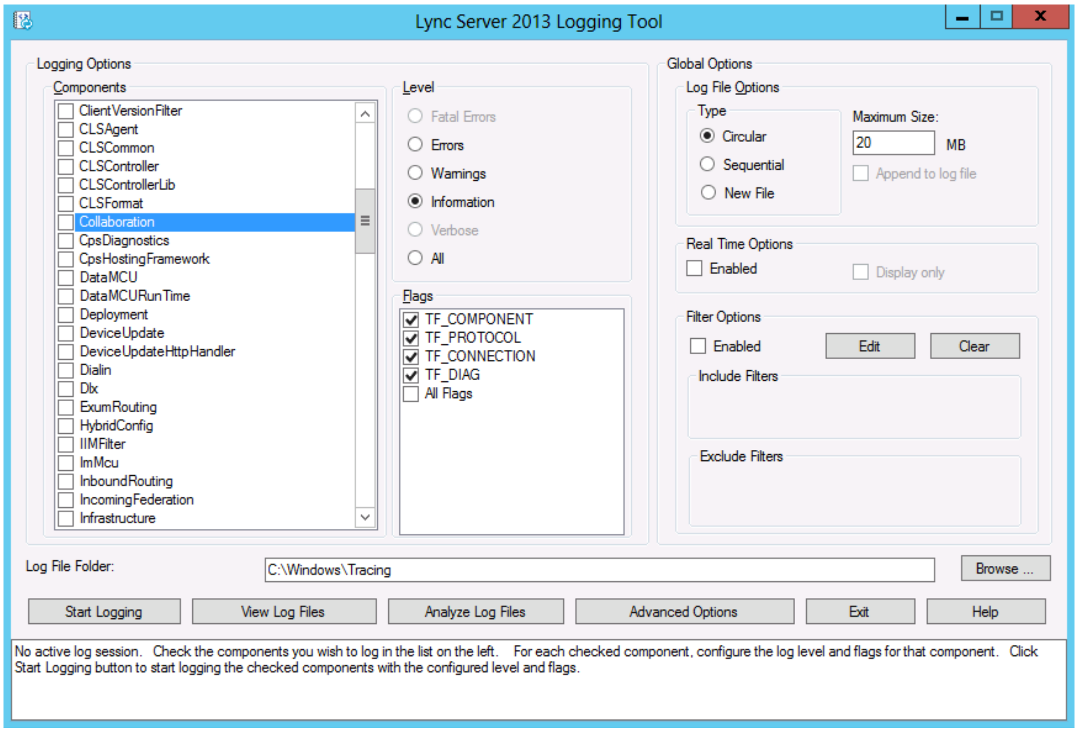
The Lync Server mediation server will listen on TCP port 5067 by default for SIP signaling. In our example we are listening on 5060 on the PSTN facing NIC. From the Standard Edition pool or Enterprise Edition pool the Mediation Server would be listening on port 5061. Tests may include:

- 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.



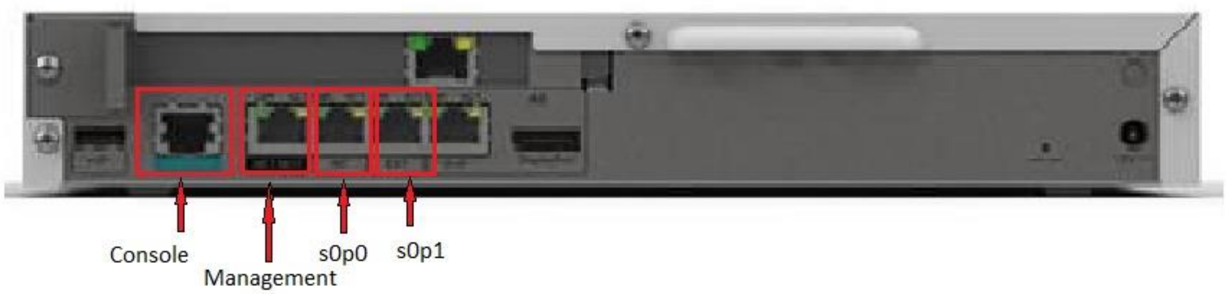
Appendix A

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.

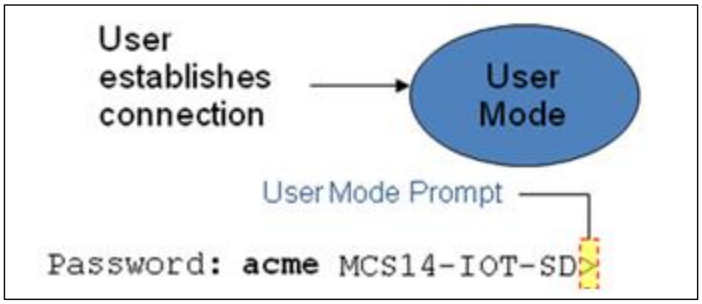


ACLI Basics

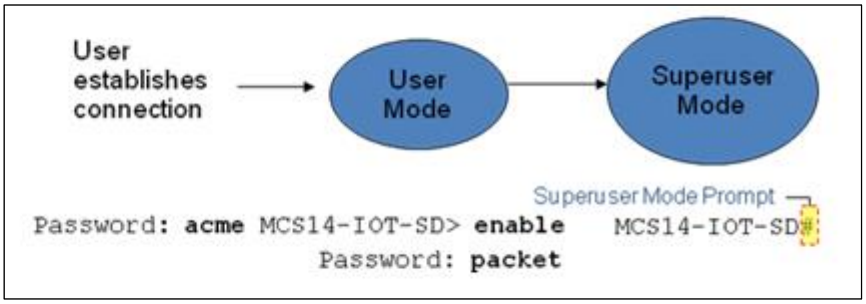
There are two password protected modes of operation within the ACLI, User mode and Superuser mode.

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.



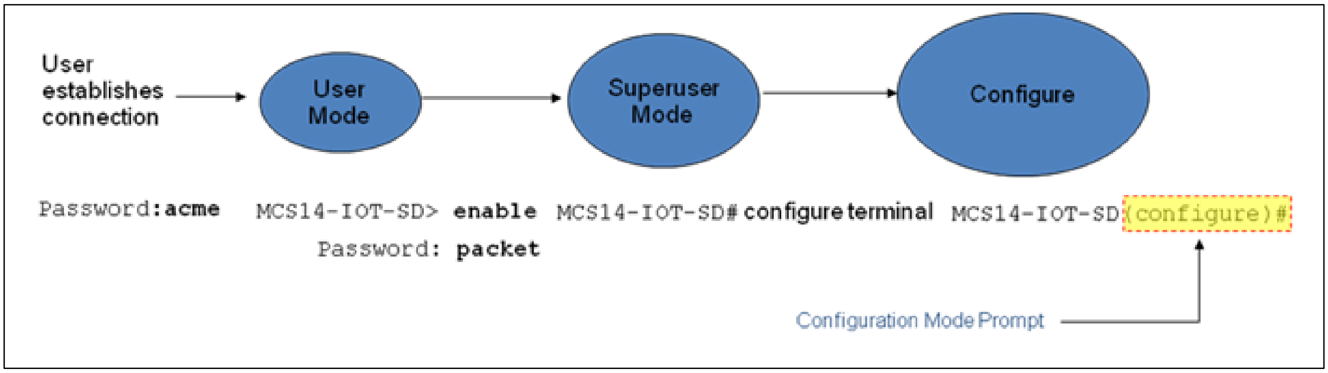
The Superuser mode allows for access to all system commands for operation, maintenance, and administration. This mode is identified by the pound sign (#) in the prompt after the target name. To enter the Superuser mode, issue the enable command in the User mode.



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.

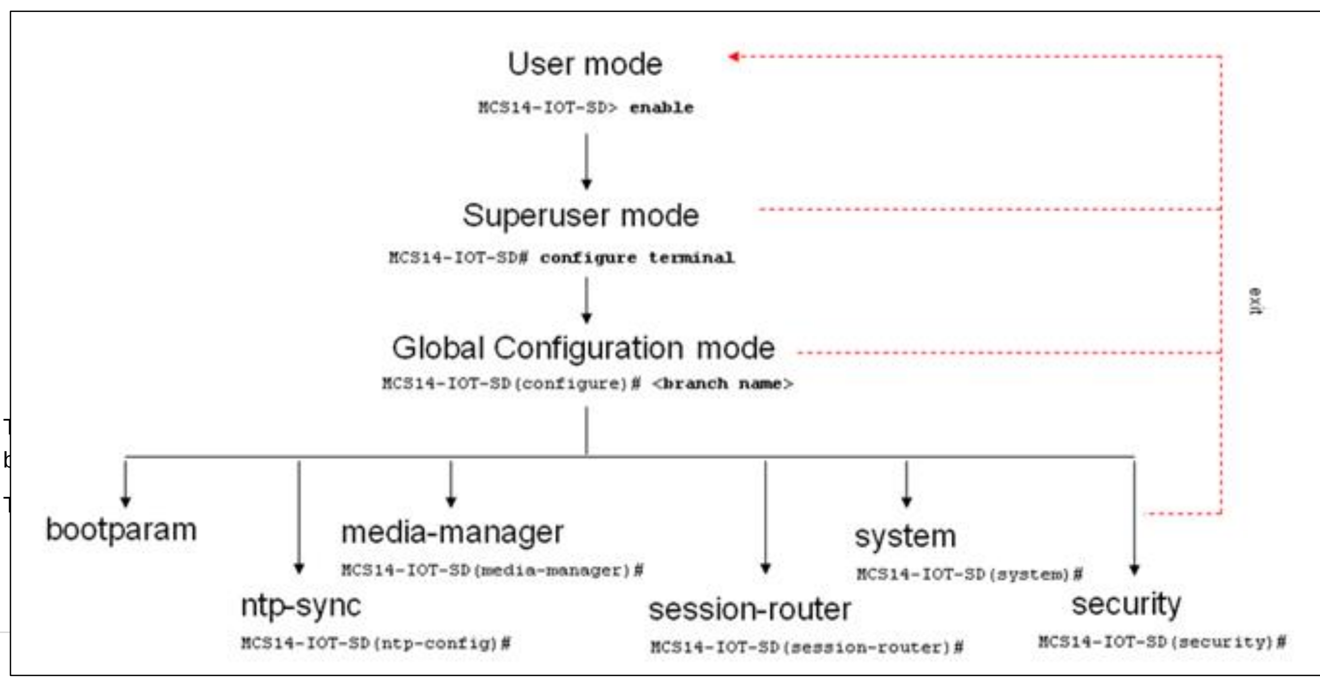
You must enter the Configuration mode to configure elements. For example, you can access the configuration branches and configuration elements for signaling and media configurations. To enter the Configuration mode, issue the `configure terminal` command in the Superuser mode.

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
boot device              : eth0
processor number         : 0
host name                :
file name                : /tffs0/nnSCX620.gz
inet on ethernet (e)     : 10.0.3.11:ffff0000
inet on backplane (b)    :
host inet (h)            : 10.0.3.100
gateway inet (g)         : 10.0.0.1
user (u)                 : anonymous
ftp password (pw) (blank = rsh)       : anonymous
flags (f)                : 0x8
target name (tn)         : MCS14-IOT-SD
startup script (s)       :
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, media-manager, 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.
2. 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 non-volatile 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.

```
oraclesbc1# activate-config
Activate-Config received, processing.
waiting 120000 for request to finish
Request to 'ACTIVATE-CONFIG' has Finished,
Activate Complete
oraclesbc1#
```

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