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Engineered to Work Together



Oracle Enterprise Session Border Controller- Acme Packet 3820 and Microsoft Lync 2013 for Enterprise SIP Trunking

Technical Application Note



Table of Contents

DOCUMENT OVERVIEW	3
INTRODUCTION	4
AUDIENCE	4
REQUIREMENTS	4
ARCHITECTURE	4
PHASE I – CONFIGURE LYNC SERVER 2013	6
REQUIREMENTS	6
ADDING THE PSTN GATEWAY	6
CONFIGURE TLS ON LYNC	18
CONFIGURING THE LYNC SERVER ROUTE.....	21
ADDITIONAL STEPS	34
PHASE II - CONFIGURE SESSION DIRECTOR	35
IN SCOPE	35
OUT OF SCOPE	35
WHAT YOU WILL NEED	35
CONFIGURATION.....	36
PHASE III – TEST CONNECTIVITY TO SIP TRUNK	82
OVERVIEW.....	82
UCOIP TEST PLAN & RESULTS	82
TROUBLESHOOTING TOOLS	91
MICROSOFT NETWORK MONITOR (NETMON)	91
WIRESHARK.....	91
EVENT VIEWER.....	91
NET-NET ESD	92
LYNC SERVER LOGGING TOOL	93
2. APPENDIX	94
KNOWN ISSUES	94



Document Overview

Microsoft Lync Server 2013 offers the ability to connect to Internet telephony service providers (ITSP) using an IP-based SIP trunk. This reduces the cost and complexity of extending an enterprise's telephony system outside its network borders. Microsoft recommends an E-SBC to provide interoperability and service assurance when connecting the Lync environment to a SIP trunk service. Acme Packet Net-Net Enterprise Session Director (Net-Net ESD) Session Border Controllers (SBCs) play an important role in SIP trunking as they are used by many ITSPs and enterprises as part of their SIP trunking infrastructure. Acme Packet solutions can also be used for enterprise Session Management applications involving Lync. In Session Management applications, the same methods described in this guide for interfacing with the Lync environment via SIP trunking apply.

This step-by-step deployment guide has been prepared as a means of ensuring that SIP trunking between Lync Server and Acme Packet SBCs is configured in the optimal manner. This guide can be used to support the SIP trunking reference topologies that are documented by Microsoft and Acme Packet in this TechNet article:

"Lync Server 2010 & OCS 2007 R2 Support for Acme Packet Session Border Controllers"
<http://blogs.technet.com/b/nexthop/archive/2011/02/21/support-for-acme-packet-session-border-controllers-in-lync-server-and-2010-communications-server-2007-r2.aspx>.

The Net-Net ESD is fully qualified by Microsoft under its Unified Communications Open Interoperability Program. It should be noted that while this deployment guide focuses on the optimal configurations for the Acme Packet Net-Net ESD SBC in a Lync Server environment, the same SBC configuration model can also be used for Microsoft OCS 2007 R2 environments. In addition, it should be noted that the Net-Net SD configuration provided in this guide focuses strictly on the Lync Server associated parameters. Many Net-Net ESD users may have additional configuration requirements that are specific to other applications. These configuration items are not covered in this guide. Please contact your Acme Packet representative for any additional information required. Additionally, the screenshots pertaining to Lync Server 2013 configuration and setup are taken to give an overview of how the setup is built and may or may not correspond to the actual configuration described elsewhere in the document.

For additional information on Lync Server, please visit the URL below:

<http://www.microsoft.com/lync>

For additional information on Acme Packet SBCs and Lync Server, please visit the URL below:

<http://www.acmepacket.com/lync>

Note: This document is to be used with Acme Packet Session Director and Enterprise Session Director platforms operating the C-series software. This includes the Net-Net 3820, 4500, and Enterprise Session Director Server Edition. For deployments involving other Acme Packet products, please contact your Acme Packet representative.

Introduction

Audience

This is a technical document intended for engineers with the purpose of configuring both the Net-Net ESD SBC and the Lync Server 2013. 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 Lync Server deployment, including Active Directory and DNS
- A dedicated Mediation Server for the SIP trunking connection
- Acme Packet Net-Net SD running software SCx6.2.0m6p1 or later

Architecture

The following reference architecture shows a logical view of the connectivity between Lync Server and the Net-Net SD.

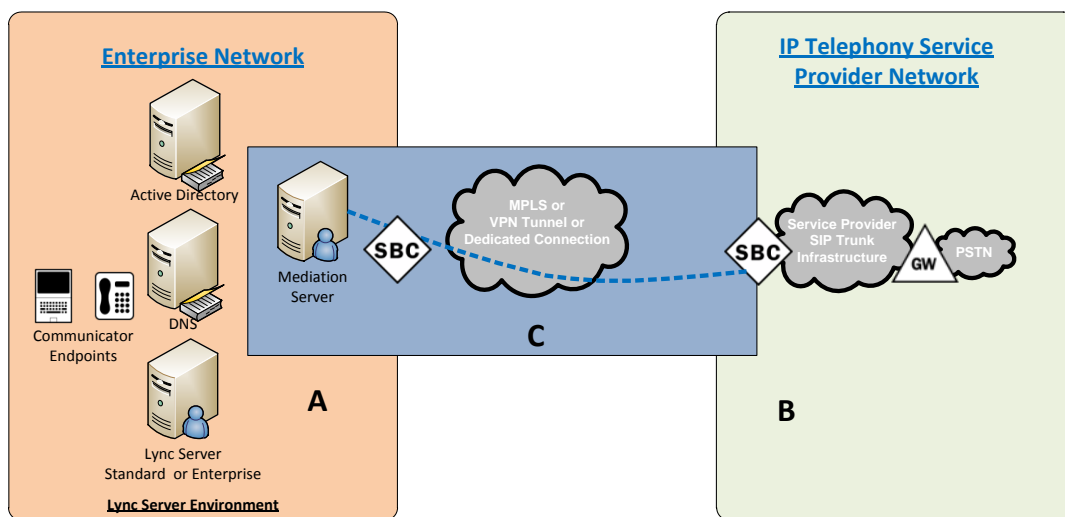


Figure 1 – Logical Reference Architecture



Area A represents the customer's on-premise infrastructure, which includes the Active Directory, DNS and Lync Server systems. Area B represents the service provider infrastructure 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 Lync Server Mediation Server and the Net-Net SD are the edge components that form the boundary of the SIP trunk. The configuration, validation and troubleshooting of the areas B and C is the focus of this document and will be described in three phases:

- Phase 1 – Configure Lync Server 2013 (define topology, pool, mediation server, add PSTN gateway and routes)
 - Phase 2 – Configure the Session Director (configure interfaces, routing, TLS/encryption)
 - Phase 3 – Test connectivity



Phase I – Configure Lync Server 2013

There are two parts for configuring Lync Server to operate with the Net-Net SD:

- Adding the Net-Net ESD as a PSTN gateway to the Lync Server infrastructure and
- Creating a route within the Lync Server infrastructure to utilize the SIP trunk connected to the Net-Net ESD

Requirements

The enterprise will have a fully functioning Lync 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.

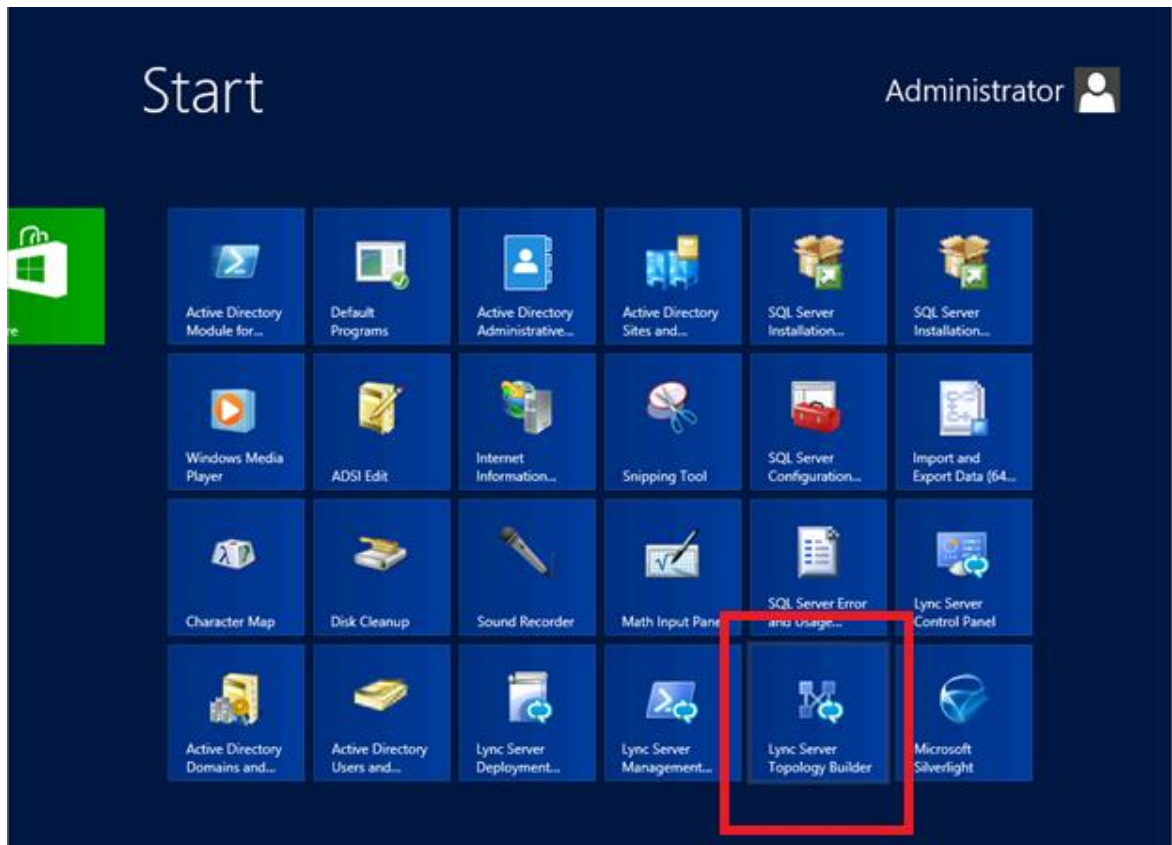
Adding the PSTN Gateway

What you will need:

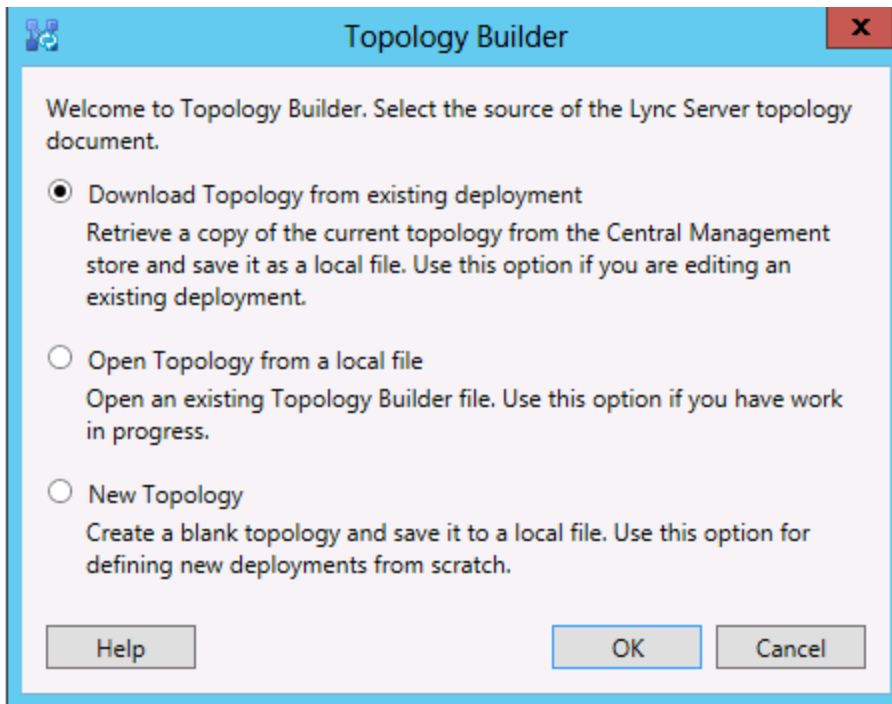
- IP address of Mediation Server external facing NIC
- IP address to be used for the Net-Net SD external facing port
- Rights to administer Lync Server Topology Builder
- Access to the Lync Server Topology Builder

Steps to add the PSTN gateway

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

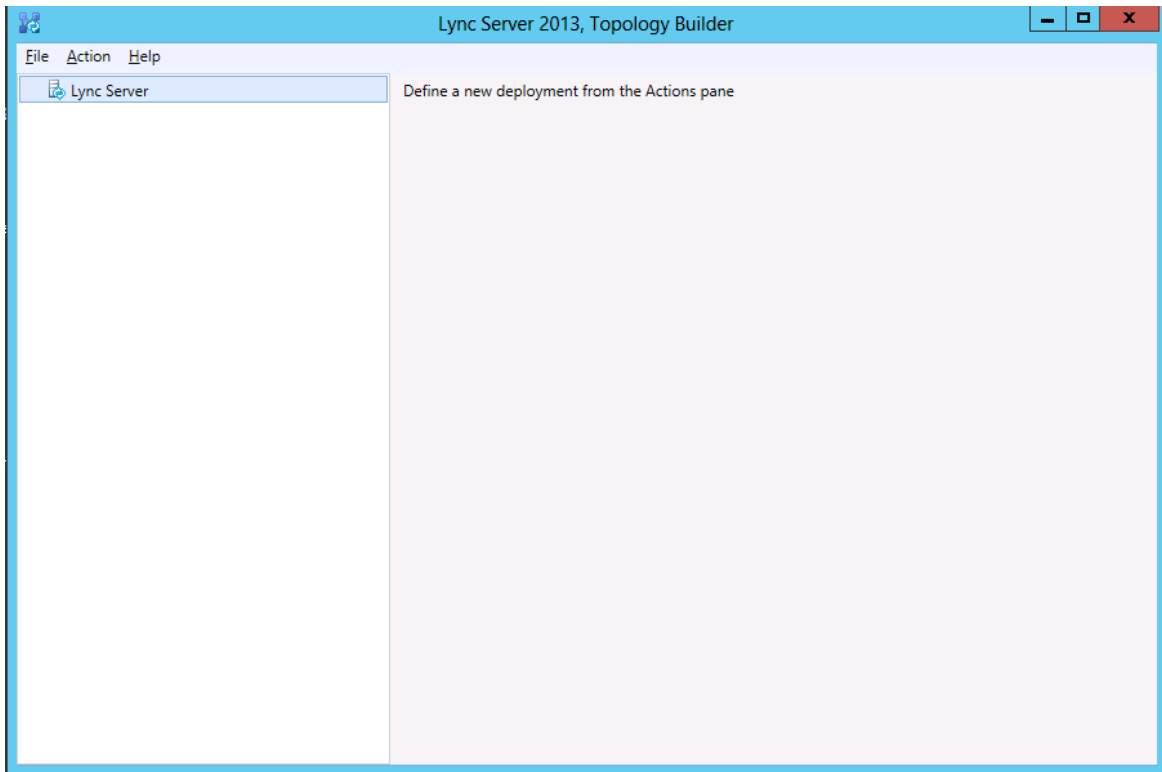


3.

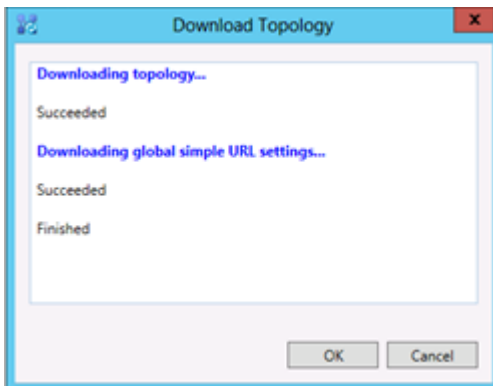


You will now be at the opening screen in the Topology Builder.

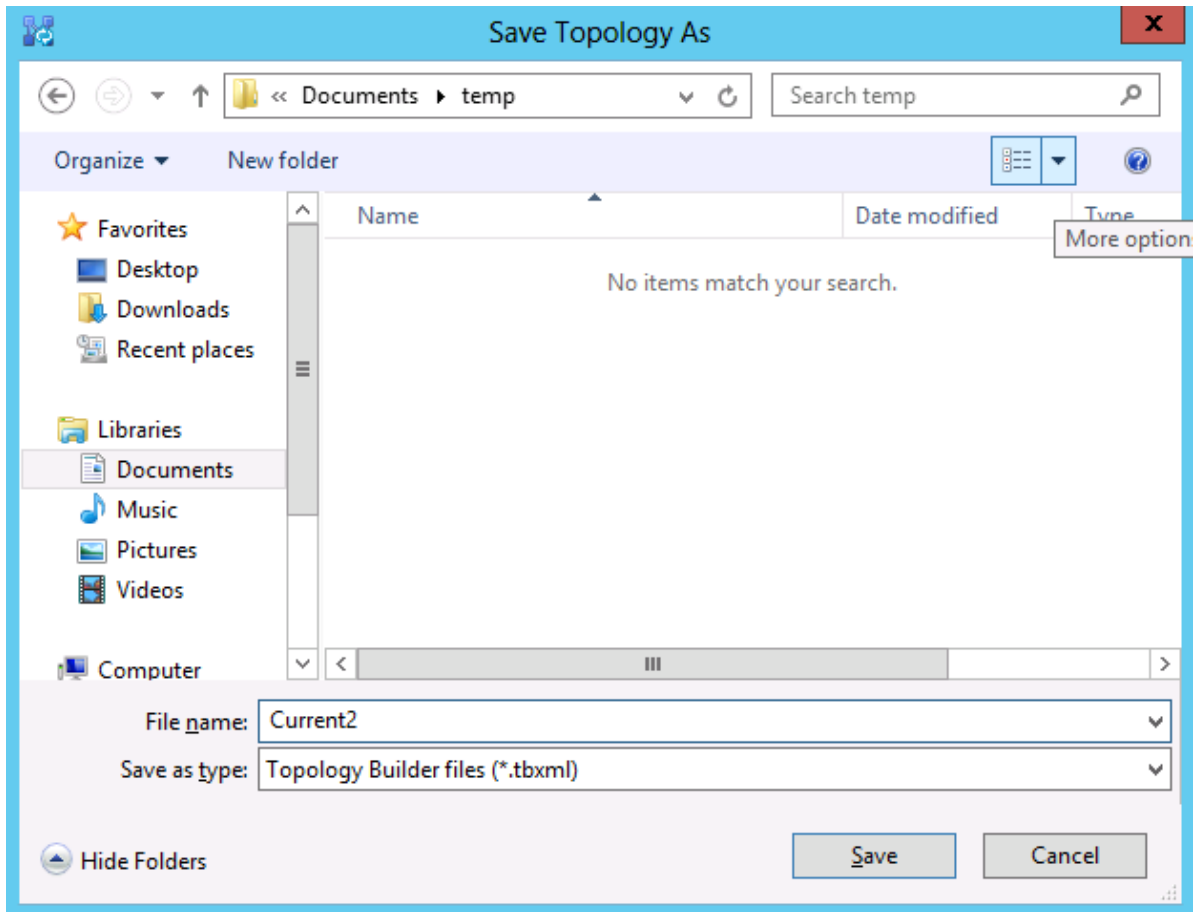
4. Click on the **Cancel** button.



5. Click on Action and select **Download Topology**.



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



- 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.
Note: This keeps track of topology changes and, if desired, will allow you to fall back from any changes you make during this installation.
- Click the **Save** button.

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

Lync Server 2013, Topology Builder

File Action Help

- Lync Server
 - Bedford

SIP domain

Default SIP domain: acmepacket.net
Additional supported SIP domains: Not configured

Simple URLs

Phone access URLs: Active Simple URL ✓

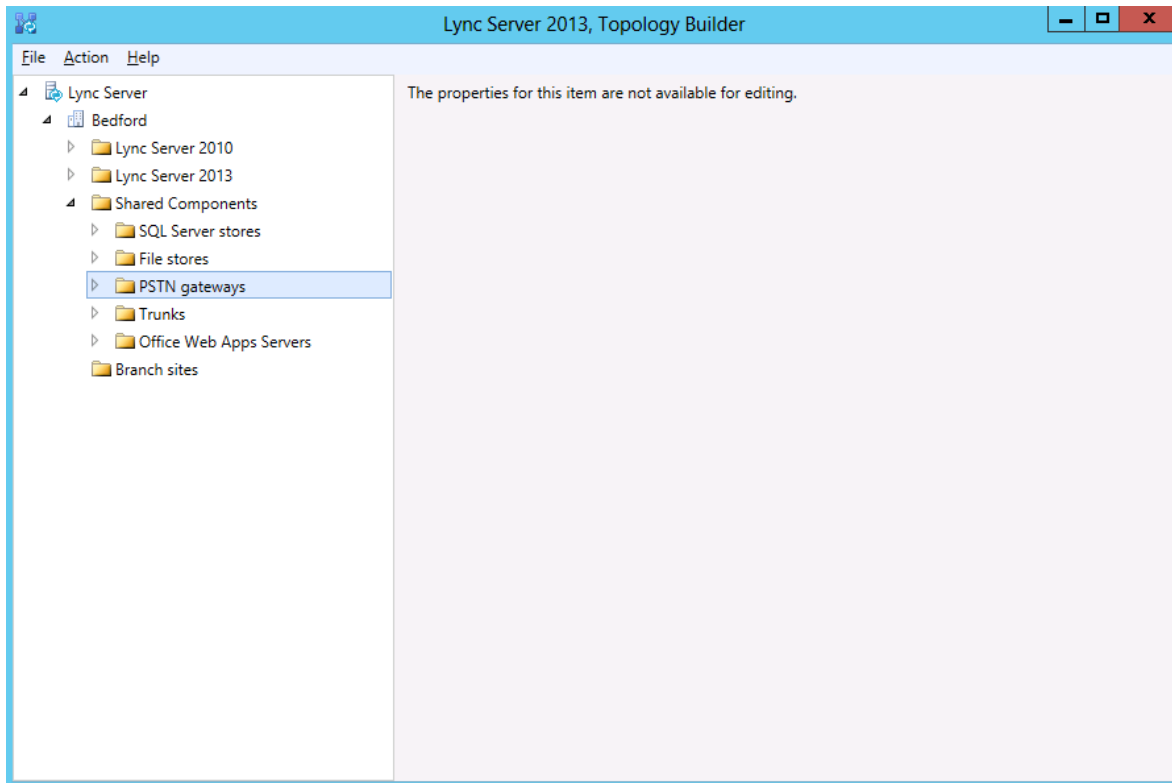
Meeting URLs: Active Simple URL SIP domain ✓

Administrative access URL: https://admin.acmepacket.net

Central Management Server

Central Management Server: Active Front End Site ✓

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



11. Right click on **PSTN Gateways** and select **New PSTN Gateway**.

Define New IP/PSTN Gateway

Define the PSTN Gateway FQDN

Define the fully qualified domain name (FQDN) for the PSTN gateway.

FQDN: *

Help Back Next Cancel

Define New IP/PSTN Gateway

Define the IP address

Enable IPv4

Use all configured IP addresses.
 Limit service usage to selected IP addresses.
PSTN IP address:

Enable IPv6

Use all configured IP addresses.
 Limit service usage to selected IP addresses.
PSTN IP address:

Help Back Next Cancel

Define New IP/PSTN Gateway

Define the root trunk

Trunk name: *
192.168.85.55

Listening port for IP/PSTN gateway: *
5060

SIP Transport Protocol:
TCP

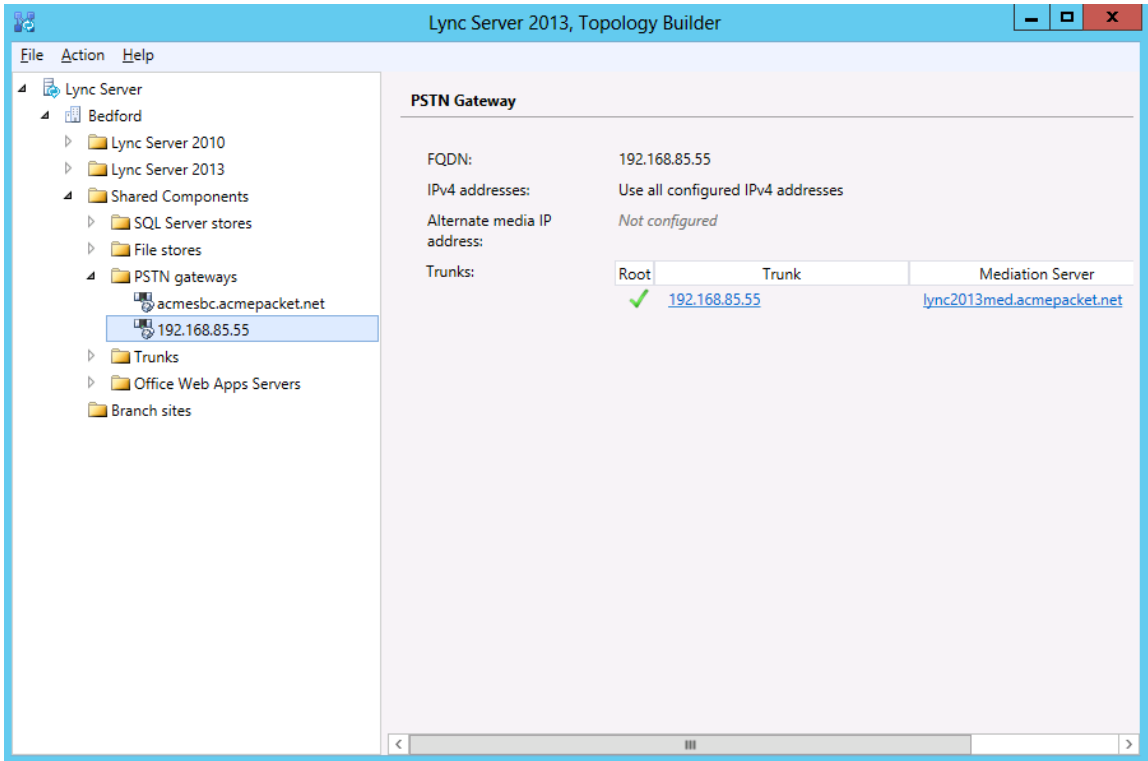
Associated Mediation Server:
lync2013med.acmepacket.net Bedford

Associated Mediation Server port: *
5068

Help Back Finish Cancel

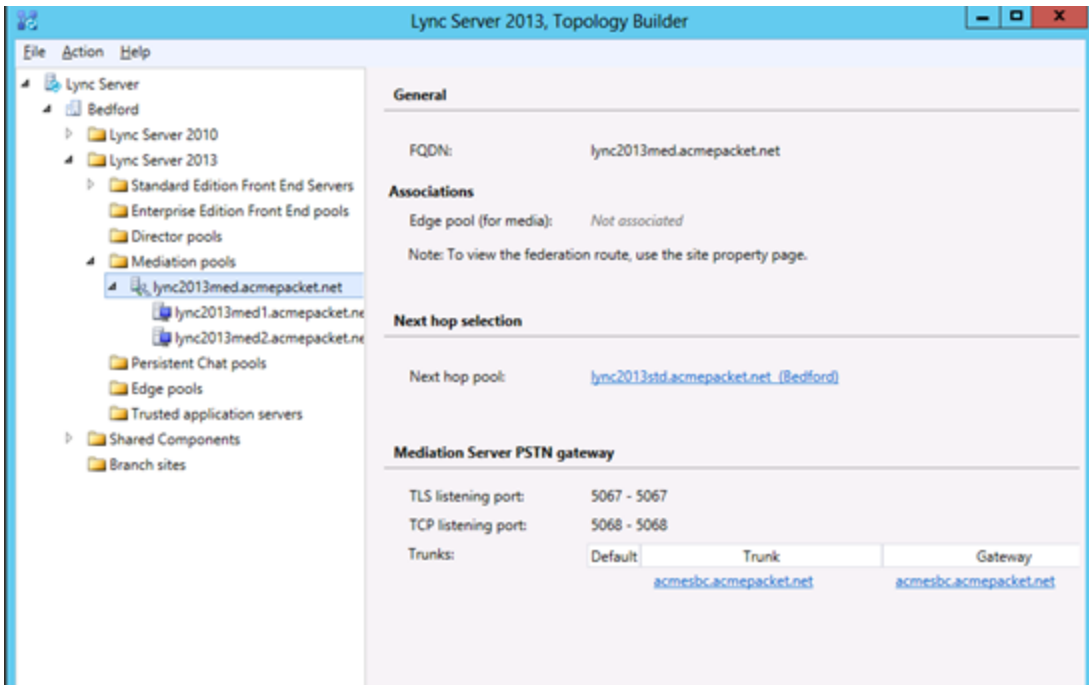
12. Enter the FQDN or the IP address that will be will be the outbound interface for the SIP Trunk on the Net-Net SD. In our example the IP address is **192.168.85.55**.
13. Enter the **Listening Port**. In our example the listening port is **5060**.
14. Select the **“Sip Transport Protocol”**. In our example it is **TCP**. Select this radio button and click **Ok**.

The PSTN Gateway for Lync Server, which is the outbound side of the Net-Net SD has now been added.



Next we will add the newly created PSTN gateway entry to the Mediation Server.

15. Expand the **Mediation Pool** list and click on the Mediation Server to be utilized. In our example the Mediation Server is **lync2013.med1.acmepackt.net**.



You will now be back at the Topology Builder screen and you can now see that your PSTN Gateway is associated with the Mediation Server

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

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



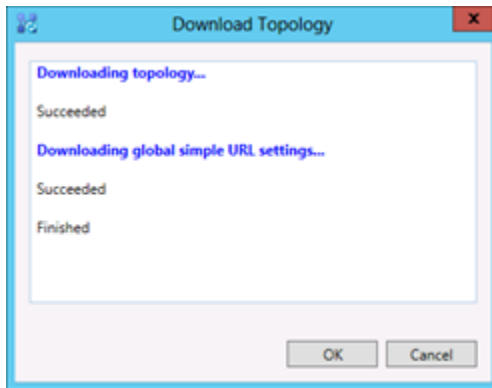
You will now be at a window showing the databases associated with site

18. Click **Next**.

When complete you should see a window from Topology Builder stating that your topology was successfully published.



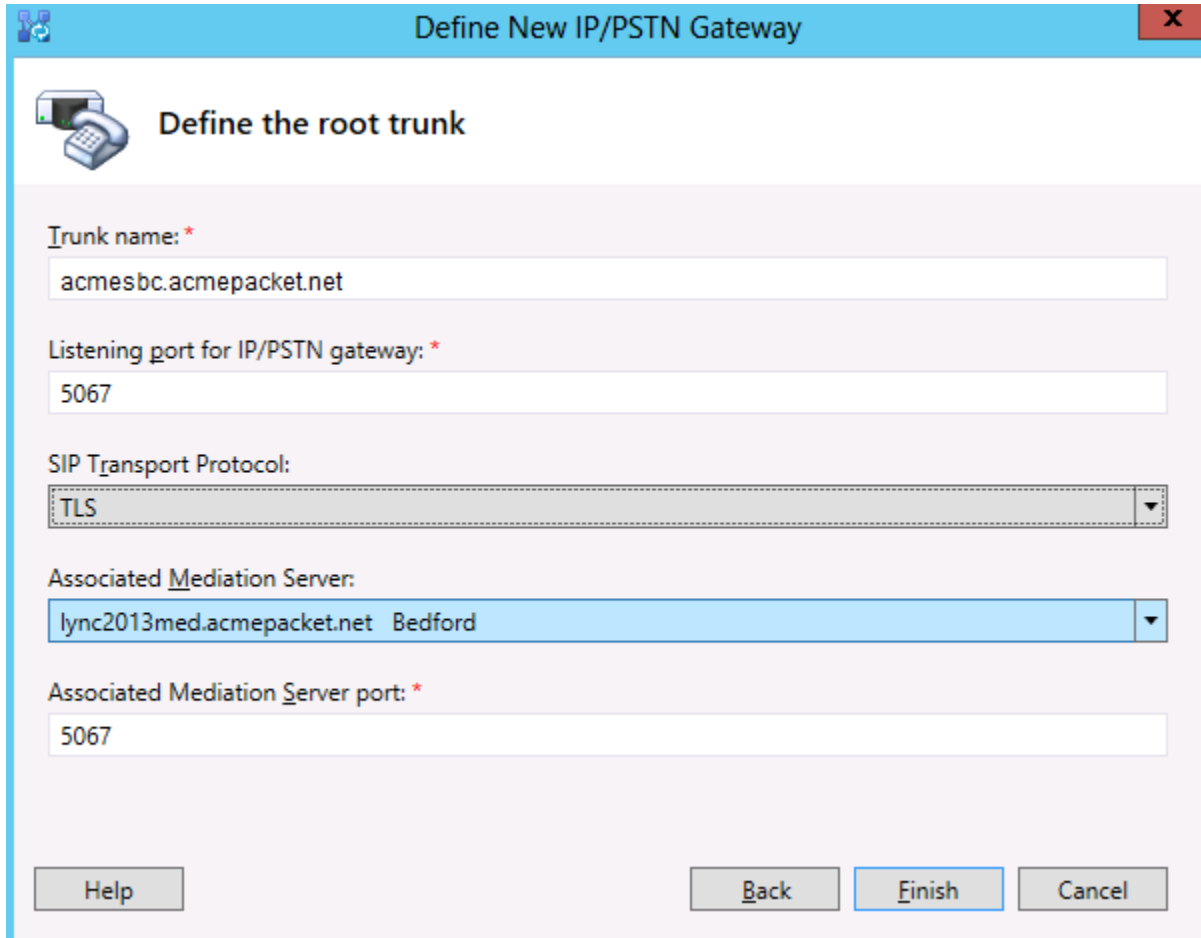
Click the **OK** button.



19. 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.

Configure TLS on Lync

1. Repeat steps from section “Adding PSTN Gateway” steps 1 thru 10. – Please NOTE: for TLS the PSTN Gateway must have a FQDN. IP Addresses are not supported.
2. Right click on **PSTN Gateways** and select **New PSTN Gateway**.



Define the root trunk

Trunk name: *

acmesbc.acmepacket.net

Listening port for IP/PSTN gateway: *

5067

SIP Transport Protocol:

TLS

Associated Mediation Server:

lync2013med.acmepacket.net Bedford

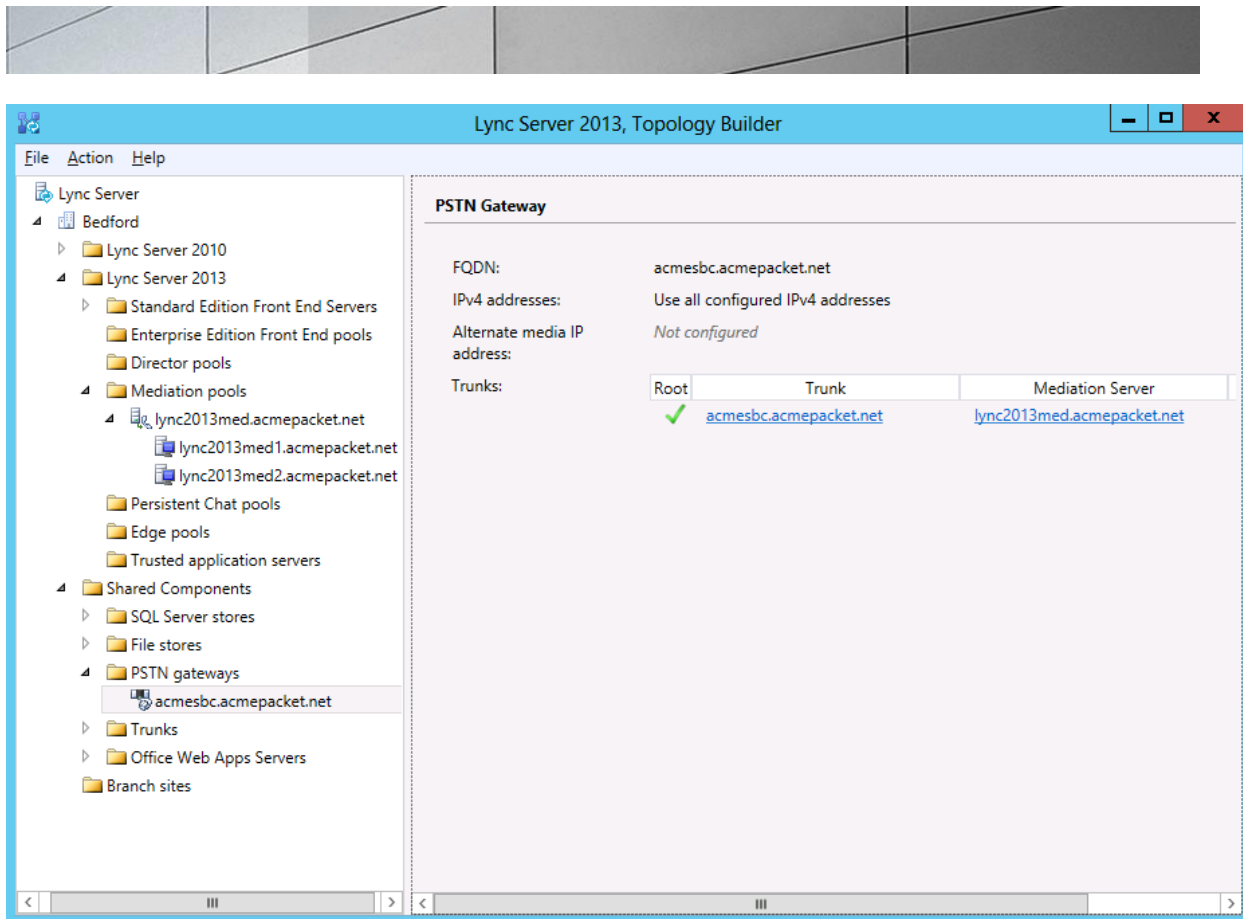
Associated Mediation Server port: *

5067

Help Back Finish Cancel

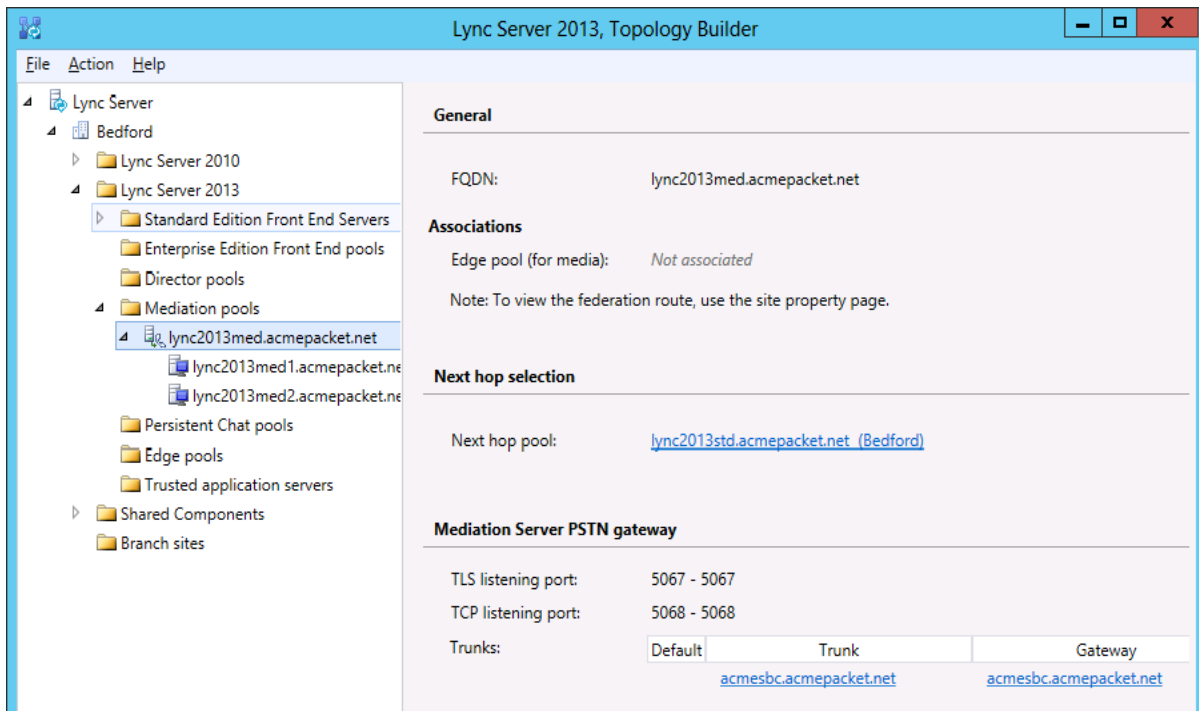
3. Enter the FQDN that will be the outbound interface for the SIP Trunk on the Net-Net ESD. In our example the FQDN is acmesbc.acmepacket.net.
4. Enter the **Listening Port**. In our example the listening port is **5067**. Mediation server as a default listens on port 5066 for TCP signaling
5. Select the “**Sip Transport Protocol**”. In our example it is **TLS**. Select this radio button and click **Finish**.

The PSTN Gateway for Lync Server, which is the outbound side of the Net-Net ESD has now been added.



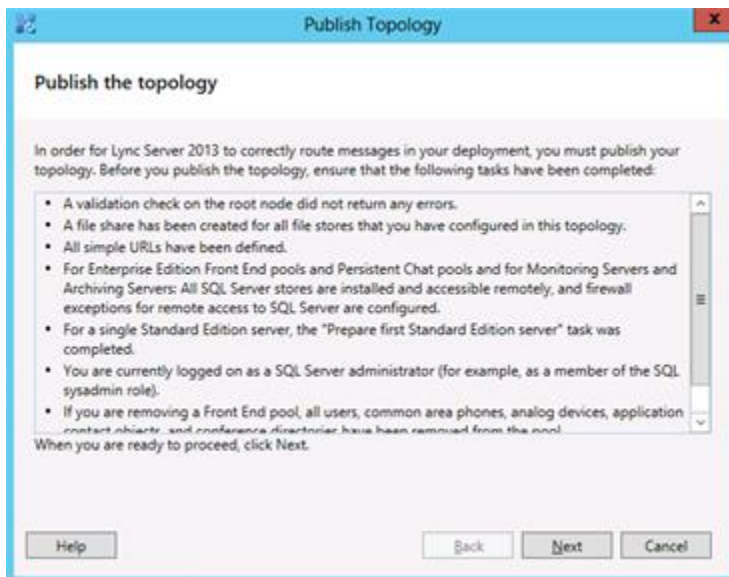
Next we will add the newly created PSTN gateway entry to the Mediation Server.

6. Expand the **Mediation Pool** list and click on the Mediation Server to be utilized. In our example the Mediation Server Pool is lync2013med.



You will now be back at the Topology Builder screen and you can now see that your PSTN Gateway is associated with the Mediation Server

7. In the upper right hand corner of your screen under **Actions** select **Topology** then select **Publish**.
8. 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



Configuring the Lync Server Route

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.

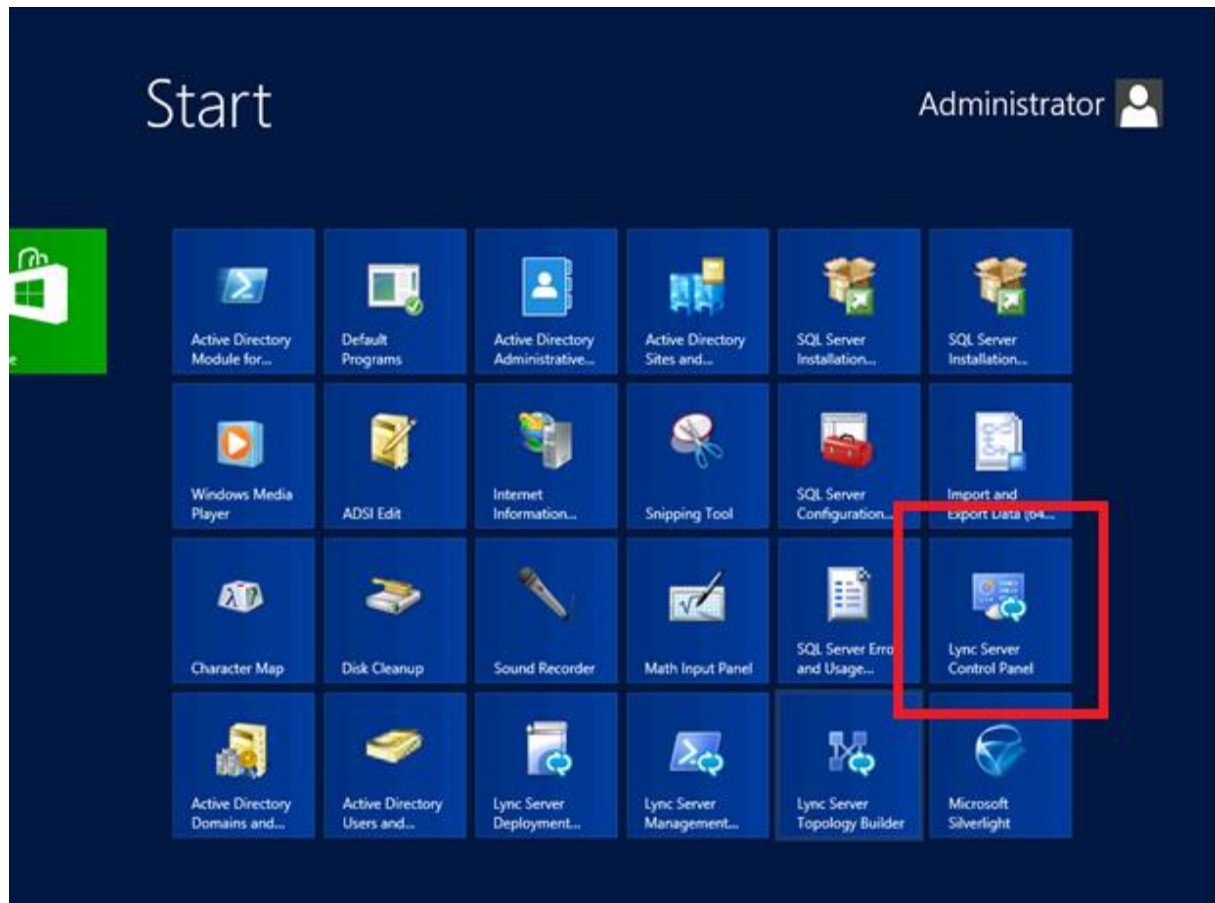
What you will need:

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

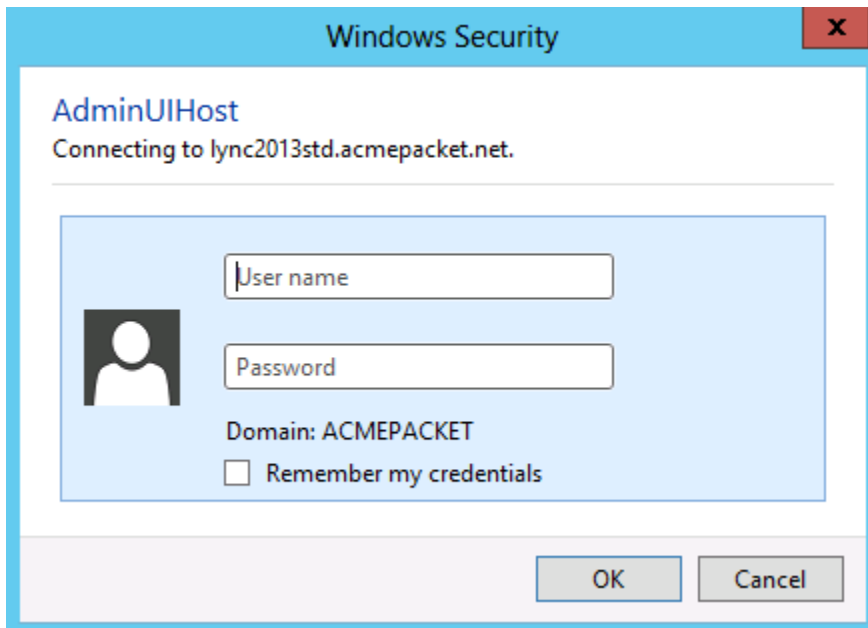
Steps to add the Lync Server Route

On the server where the CSCP is located start the console.

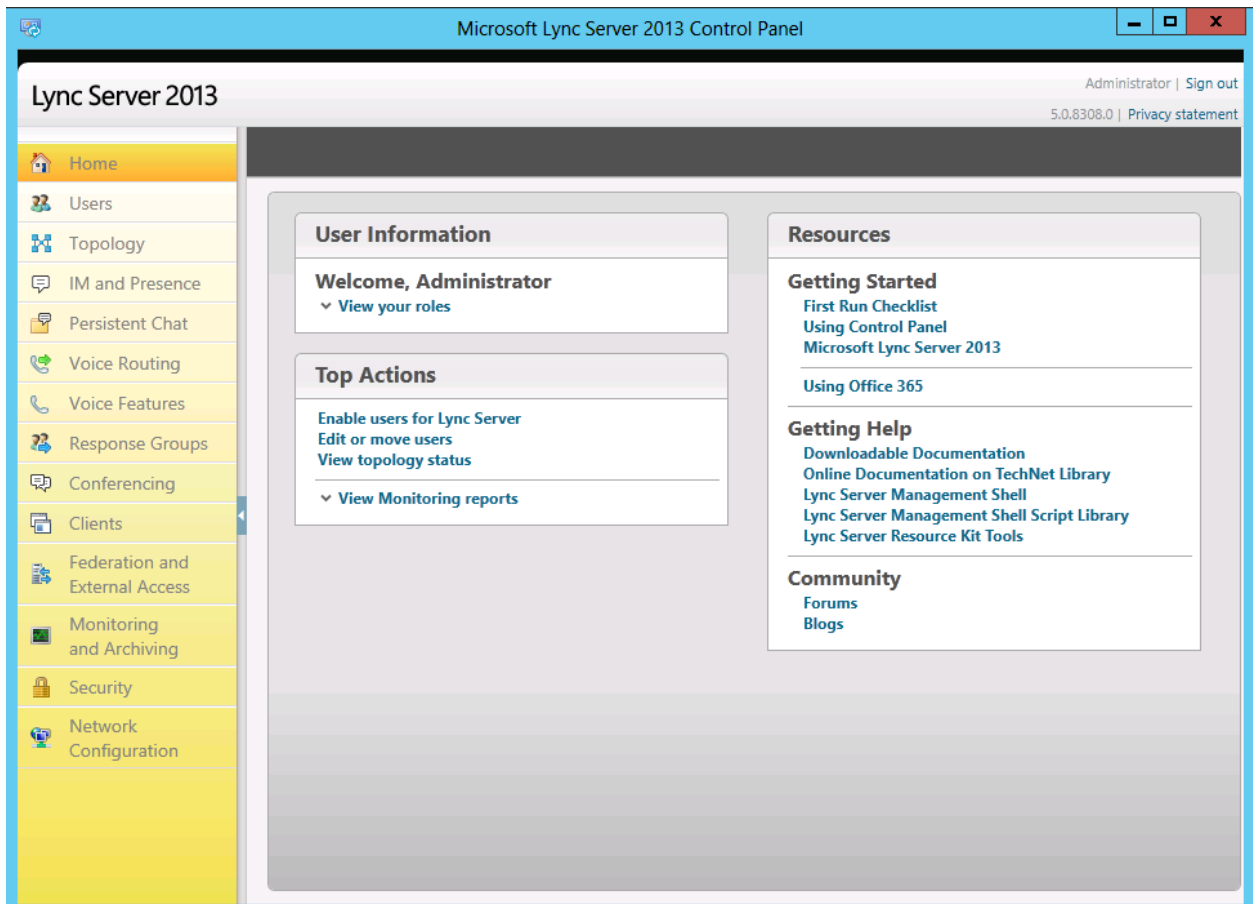
1. Click **Start**, select **All Programs**, then select **Communications Server Control Panel**



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



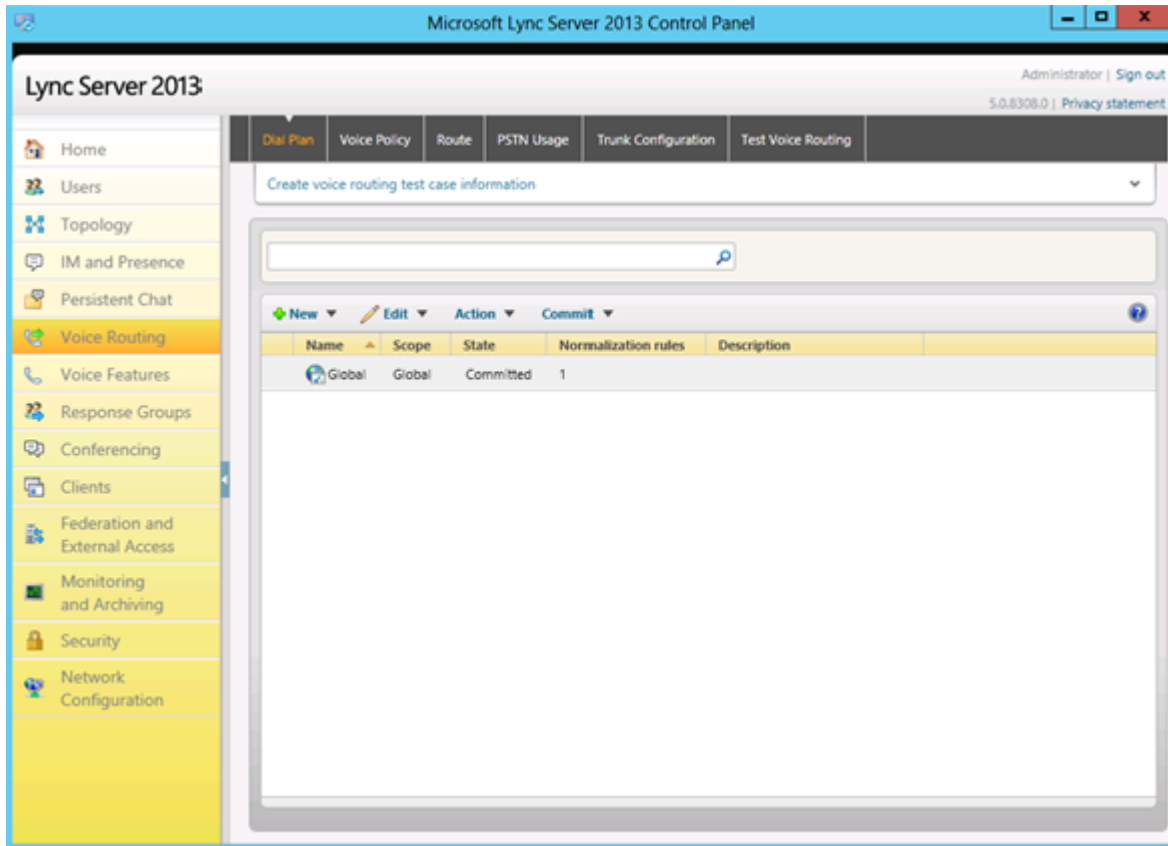
2. Once logged on, you will now be at the CSCP "Welcome Screen".



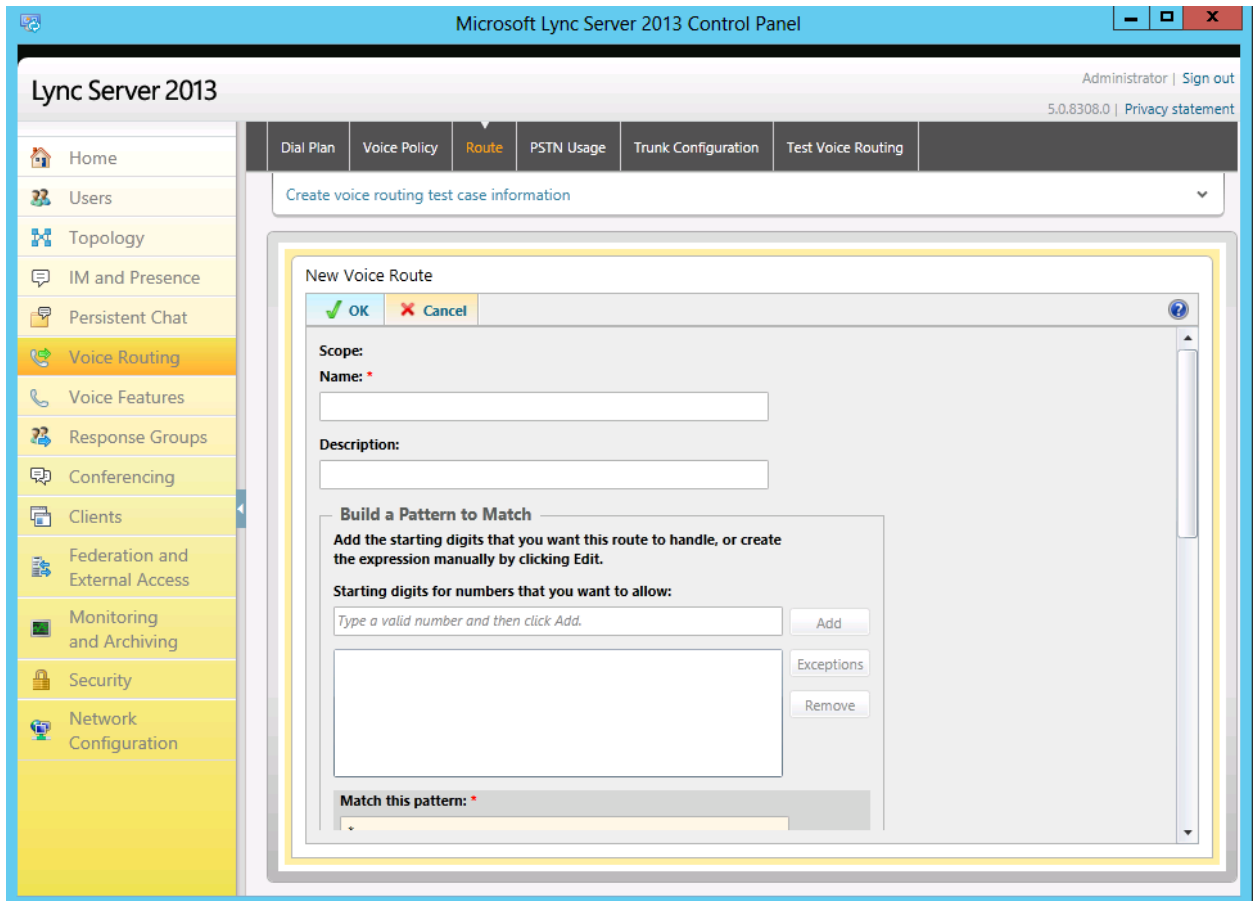
3. On the left hand side of the window click on **Voice Routing**.

You will now be in the Voice Routing section of the CSCP.

4. On the top row of tabs select **Route**.

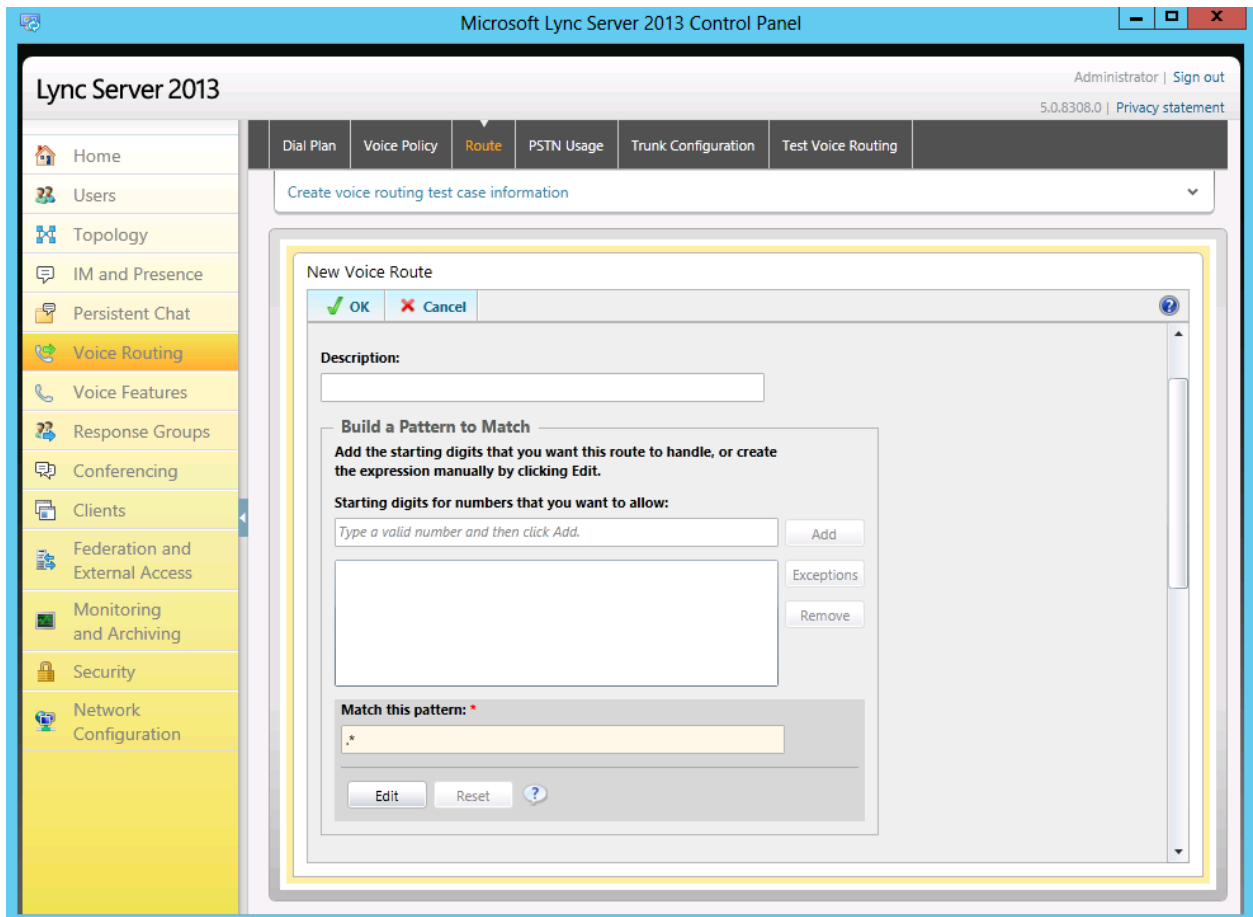


5. On the content area toolbar, click **+New**.

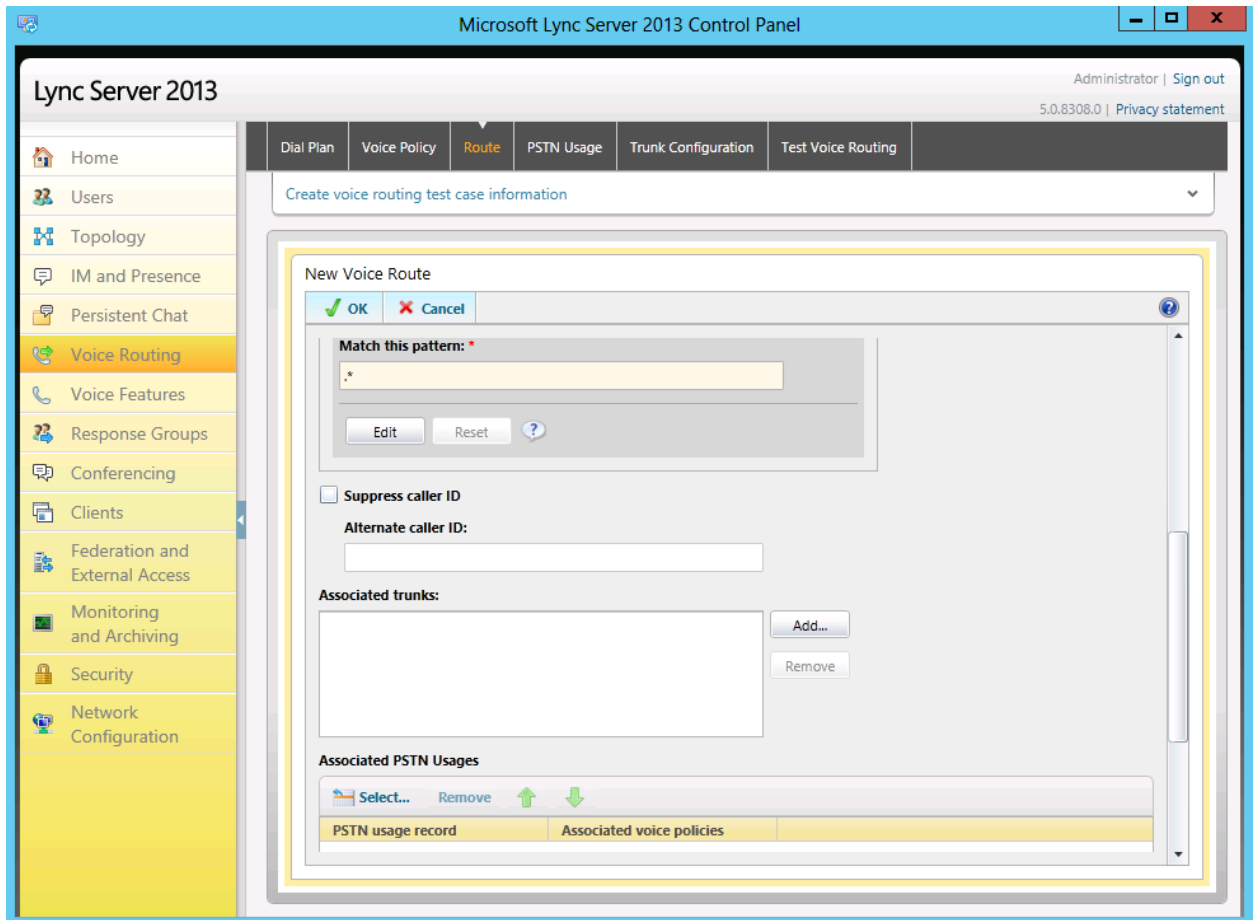


6. On the Create Voice Route page, in the Name field, enter the name you have selected for the Route. In our example, it is **Test**.

7. Next you build a Pattern Match for the phone numbers you want this route to handle. In our example we use “.*” since we were using a very simple dial plan for this route and wish to match any outgoing call.



- Next you want to associate the Voice Route with the PSTN gateway you have just created scroll down to Associated Trunks, click on the **Add** button.



You will now be at a window showing available PSTN Gateways to associate your Voice Route.

Select Trunk

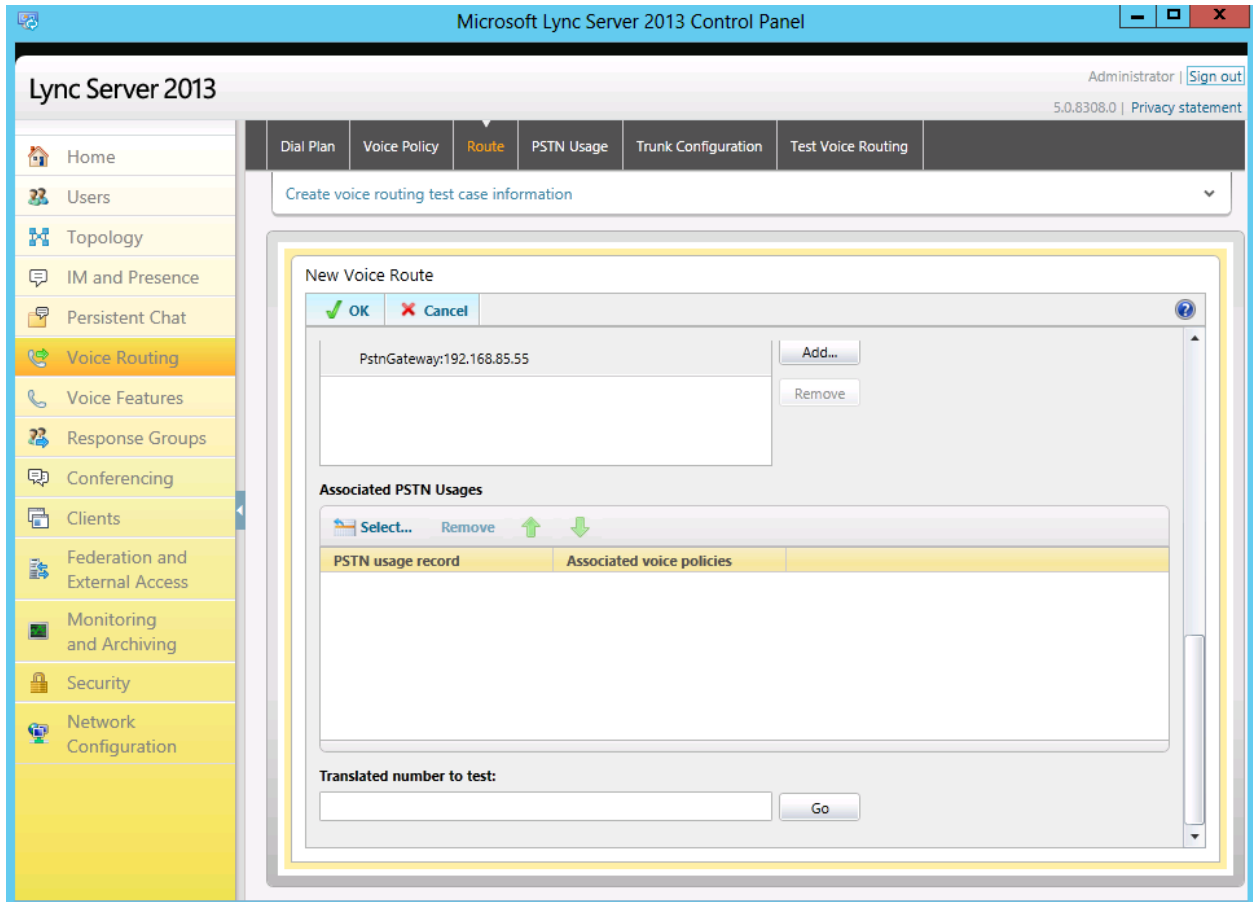


Service	Site
PstnGateway:acmesbc.acmepa...	Bedford
PstnGateway:192.168.85.55	Bedford

OK

Cancel

9. Click on the PSTN gateway that you just created and then click the **OK** button.



You can now see that you have associated your PSTN gateway with the route you created.

Note that the **Suppress Caller ID**: allows the manipulation of caller ID information for outbound calls, in order to mask employees' direct-dial extensions and replace them with the generic corporate or departmental numbers, this is not a necessary step for this installation, but may need to be addressed by customer policy.

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.

10. Click on the **Select** button under "Associated PSTN Usages".



Select PSTN Usage Record X

🔍

PSTN usage record name	Associated routes	Associated voice policies
Internal		
Local		
Long Distance		
Test		Test

11. Select the appropriate PSTN Usage Record then click the **OK** button.

New Voice Route

Resource gateway

PstnGateway:192.168.85.55

Associated PSTN Usages

PSTN usage record	Associated voice policies
Test	<input type="button" value="Test"/>

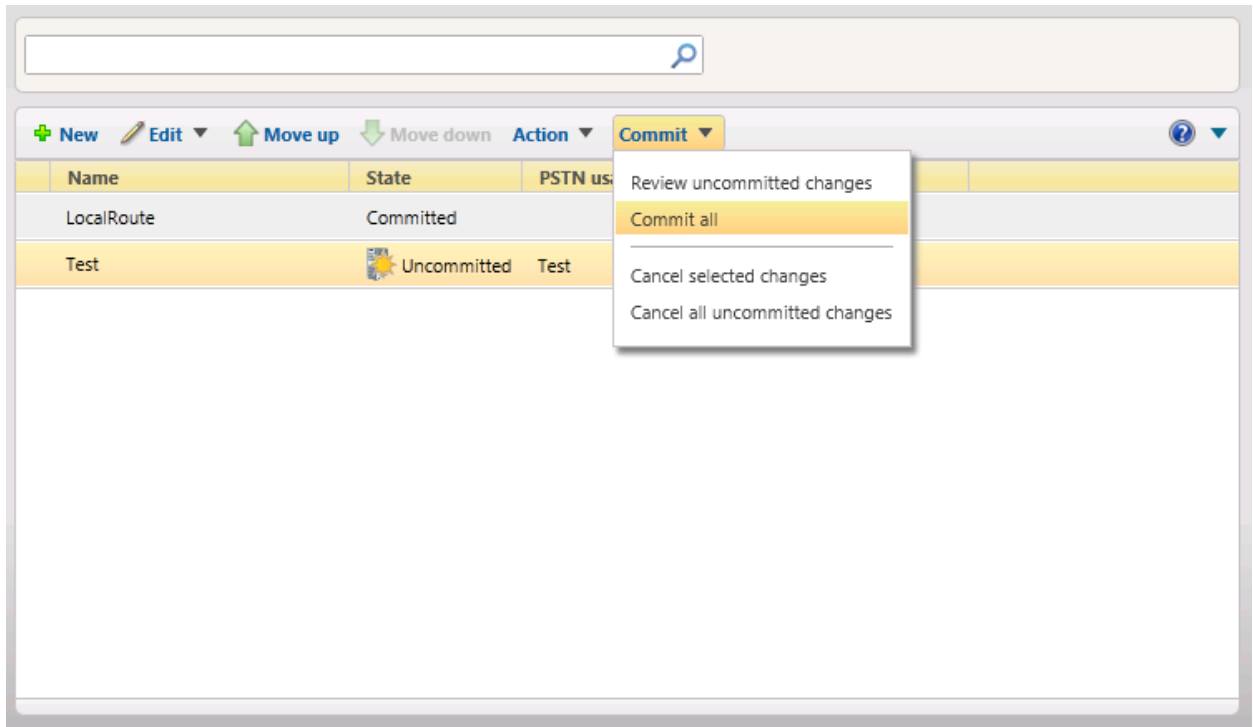
12. You will now see the Associated PSTN Gateway Usages which you have added. Click the **OK** button at the top New Voice Route screen.

192.85

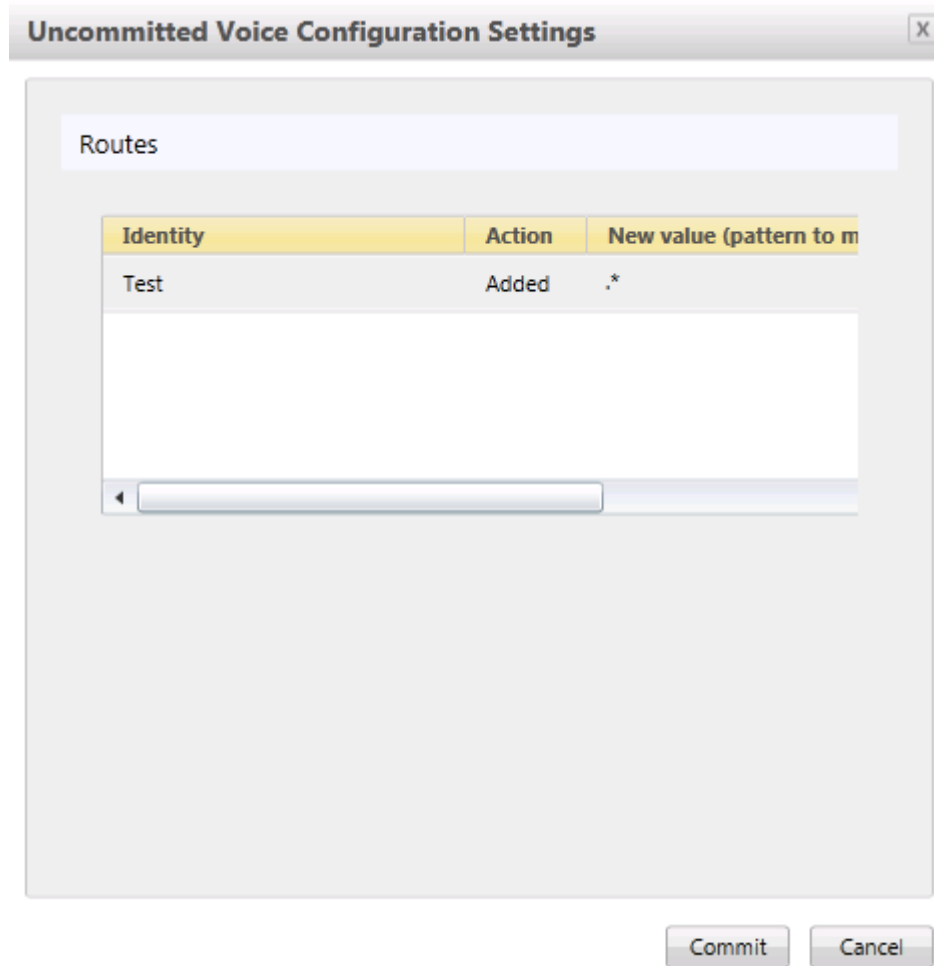
The screenshot displays a network configuration interface with a sidebar on the left and a main content area on the right. The sidebar contains various configuration categories, with 'Voice Routing' highlighted in yellow. The main content area has a top navigation bar with tabs for 'Dial Plan', 'Voice Policy', 'Route', 'PSTN Usage', 'Trunk Configuration', and 'Test Voice Routing'. Below the navigation bar is a search bar and a toolbar with options: 'New', 'Edit', 'Move up', 'Move down', 'Action', and 'Commit'. A table below the toolbar lists PSTN Usage entries:

Name	State	PSTN usage	Pattern to match
global	Committed	global	*
new test	Uncommitted		^14255775035

13. Click the **Commit** drop-down menu, and then **Commit All**.



14. On the Uncommitted Voice Configuration Settings window, click **Commit**.





15. On the **Lync Server Control Panel** prompt, click **OK**.

16. 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; and
- PSTN usage policies.

To go through them all is out of scope for this document.



Phase II - Configure Session Director

In this section we describe the steps for configuring a Net-Net SD for use with Lync Server in a SIP trunking scenario.

In Scope

The following Step-by-Step guide configuring the Net-Net ESD assumes that this is a newly deployed device dedicated to a single customer.

Note that Acme Packet offers several products and solutions that can interface with Lync Server. This document covers the setup for the Net-Net SD platforms software S-Cx 6.2.0m6p1 or later. A Net-Net 3800-series (NN3820) platform was used as the platform for developing this guide. If instructions are needed for other Acme Packet products, please contact your Acme Packet representative.

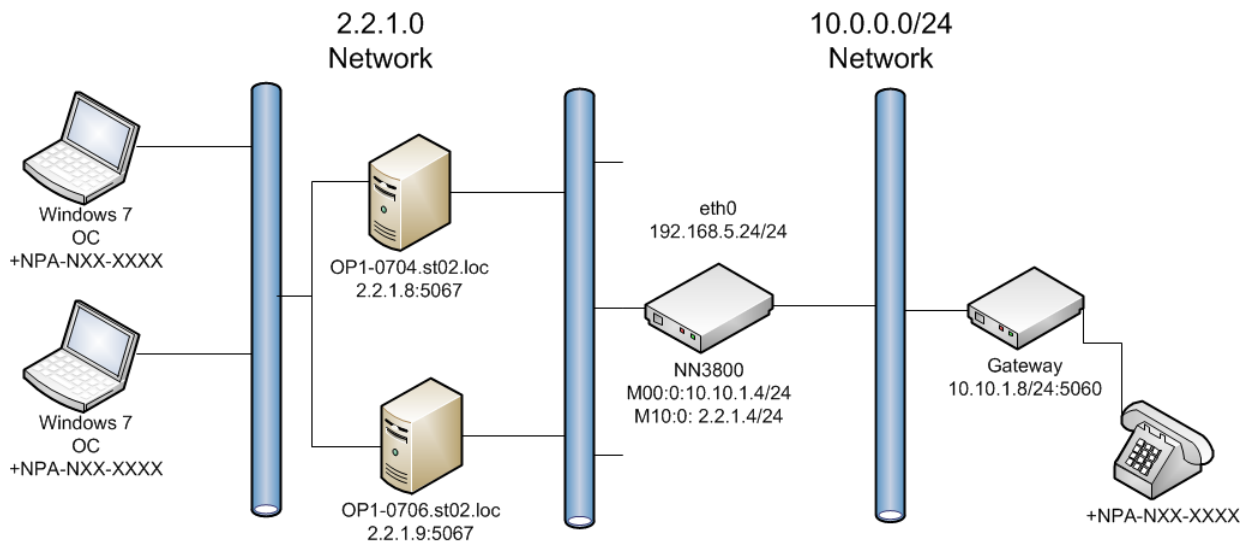
Out of Scope

- Configuration of Network management including SNMP and RADIUS; and
- Redundancy configuration

What you will need

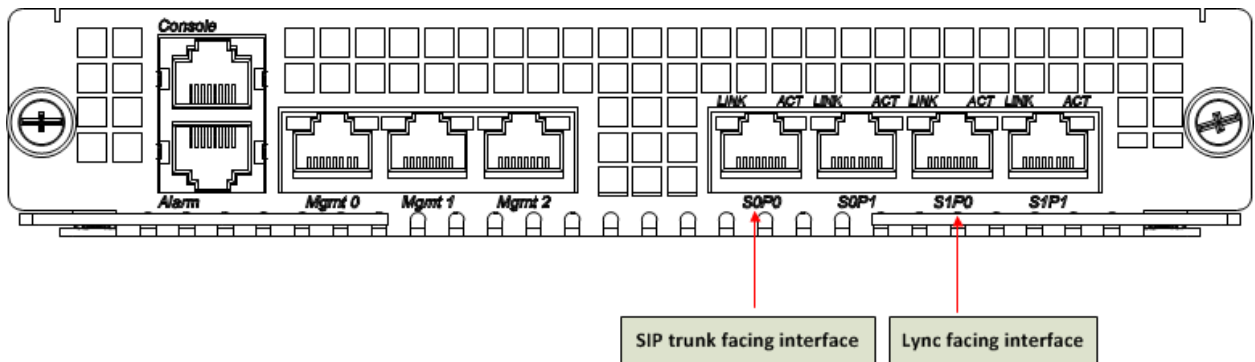
- Serial Console cross over cable with RJ-45 connector
- Terminal emulation application such as PuTTY or HyperTerm
- Passwords for the User and Superuser modes on the Net-Net SD
- Signaling IP address and port of Lync Mediation Server
- Signaling and media IP addresses and ports to be used on the Net-Net SD facing Lync and service provider SIP trunk
- Signaling IP address and port of the next hop network element in the service provider SIP trunk network
- IP address of the enterprise DNS server

Lync Server 2010 Acme Packet Test Topology



Configuration

Once the Net-Net SD is racked and the power cable connected, you are ready to set up physical network connectivity.



Plug the slot 0 port 0 (s0p0) interface into your SIP trunk provider (SIP trunk facing) network and the slot 0 port 1 (s1p0) interface into your Lync (Lync mediation server-facing) network as shown in the diagram above. Once connected, perform you are ready to power on and perform the following steps.

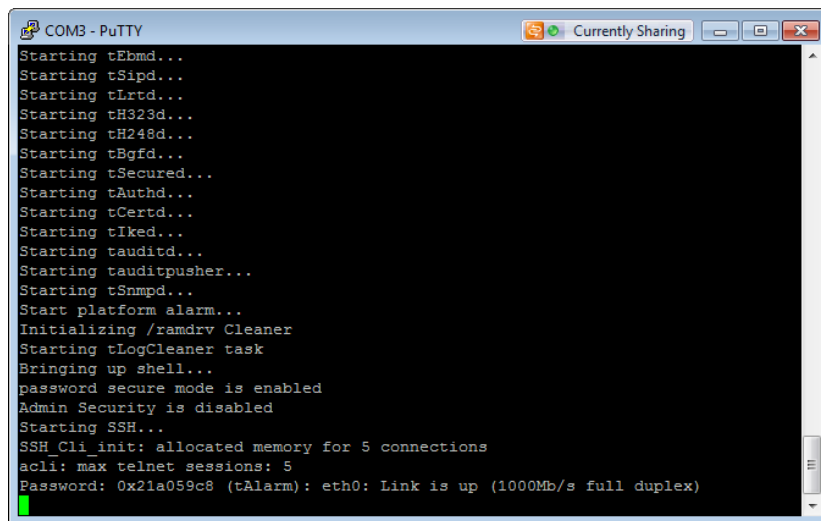
All commands are in bold, such as **configure terminal**; parameters in bold red such as **ACME1A** are parameters which are specific to an individual deployment. **Note:** The ACLI is case sensitive.

1. Establish the serial connection to the Net-Net SD.

Confirm the Net-Net SD is powered off and connect the serial console cable to the Net-Net SD to 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

Start the Net-Net SD and confirm that you see the following output from the bootup sequence.



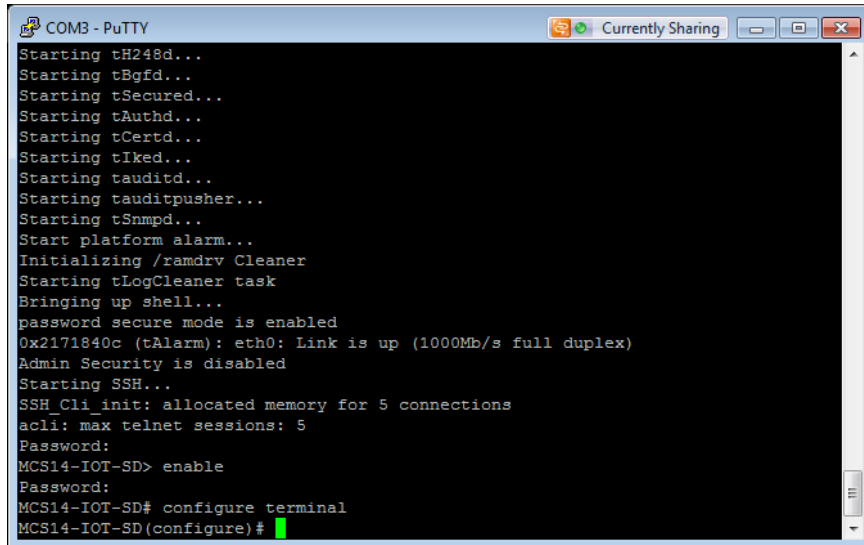
```
COM3 - PuTTY
Starting tEcmd...
Starting tSipd...
Starting tLrtd...
Starting tH323d...
Starting tH248d...
Starting tBgfd...
Starting tSecured...
Starting tAuthd...
Starting tCerte...
Starting tIked...
Starting tauditd...
Starting tauditpusher...
Starting tSnmpd...
Start platform alarm...
Initializing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

2. Login to the Net-Net SD and enter the configuration mode

Enter the following commands to login to the Net-Net SD and move to the configuration mode. Note that the default Net-Net SD password is “**acme**” and the default super user password is “**packet**”.

```
Password: acme
ACME1A> enable
Password: packet
ACME1A# configure terminal
ACME1A (configure)#
```

You are now in the Global Configuration mode.



```
COMB - PuTTY
Starting tH248d...
Starting tBgfd...
Starting tSecured...
Starting tAuthd...
Starting tCertd...
Starting tIked...
Starting tauditd...
Starting tauditpusher...
Starting tSnmpd...
Start platform alarm...
Initializing /ramdrv Cleaner
Starting tLogCleaner task
Bringing up shell...
password secure mode is enabled
0x2171840c (tAlarm): eth0: Link is up (1000Mb/s full duplex)
Admin Security is disabled
Starting SSH...
SSH_Cli_init: allocated memory for 5 connections
acli: max telnet sessions: 5
Password:
MCS14-IOT-SD> enable
Password:
MCS14-IOT-SD# configure terminal
MCS14-IOT-SD(configure)#
```

3. Configure system element values

To configure system element values, use the **system-config** command under the system branch. Then enter values appropriate to your environment, including your default gateway IP address for your management Ethernet interface.

```
ACME1A (configure)# system
ACME1A (system)# system-config
ACME1A (system-config)# hostname ACME1A
ACME1A (system-config)# description "Lync Server 2013 SIP Trunking "
ACME1A (system-config)# location "Redmond, WA"
ACME1A (system-config)# mib-system-contact "brian@cs.loc"
ACME1A (system-config)# default-gateway 10.176.32.1
ACME1A (system-config)# done
```

Once the **system-config** settings have completed and you enter **done**, the Net-Net SD will output a complete listing of all current settings. This will apply throughout the rest of the configuration and is a function of the **done** command. Confirm the output reflects the values you just entered as well as any configuration defaults.

```
system-config
hostname
description                Lync Server 2013 SIP Trunking
location                    Redmond, WA
mib-system-contact
mib-system-name
mib-system-location        Redmond, WA
snmp-enabled                enabled
enable-snmp-auth-traps     disabled
enable-snmp-syslog-notify  disabled
```

enable-snmp-monitor-traps	disabled
enable-env-monitor-traps	disabled
snmp-syslog-his-table-length	1
snmp-syslog-level	WARNING
system-log-level	WARNING
process-log-level	NOTICE
process-log-ip-address	0.0.0.0
process-log-port	0
collect	
sample-interval	5
push-interval	15
boot-state	disabled
start-time	now
end-time	never
red-collect-state	disabled
red-max-trans	1000
red-sync-start-time	5000
red-sync-comp-time	1000
push-success-trap-state	disabled
call-trace	disabled
internal-trace	disabled
log-filter	all
default-gateway	10.176.32.1
restart	enabled
exceptions	
telnet-timeout	0
console-timeout	0
remote-control	enabled
cli-audit-trail	enabled
link-redundancy-state	disabled
source-routing	disabled
cli-more	disabled
terminal-height	24
debug-timeout	0
trap-event-lifetime	0
default-v6-gateway	::
ipv6-support	disabled

4. Configure Physical Interface values

To configure physical Interface values, use the **phy-interface** command under the system branch. To enter the system branch from system-config, you issue the **exit** command then the **phy-interface** command.

You will first configure the slot 0, port 0 interface designated with the name s0p0 on the rear of the system. This will be the interface/port that connects to your SIP trunk provider.

```
ACME1A(system-config)# exit
ACME1A(system-interface)# phy-interface
ACME1A(phy-interface)# name M00
ACME1A(phy-interface)# operation-type media
ACME1A(phy-interface)# slot 0
ACME1A(phy-interface)# port 0
ACME1A(phy-interface)# done
```

Once the **phy-interface** settings have completed for slot 0 port 0 and you enter **done**, the Net-Net ESD will output a complete listing of all current settings. Confirm the output reflects the values you just entered.

```
phy-interface
name                M00
operation-type      Media
port                0
slot                0
virtual-mac
admin-state         enabled
auto-negotiation    enabled
duplex-mode         FULL
speed               100
overload-protection disabled
```

You will now configure the slot 1 port 0 phy-interface, specifying the appropriate values. This will be the interface/port that connects to the Lync Mediation server (lync facing) network.

```
ACME1A(phy-interface)# name M10
ACME1A(phy-interface)# operation-type media
ACME1A(phy-interface)# slot 1
ACME1A(phy-interface)# port 0
ACME1A(phy-interface)# done
```

```
phy-interface
name                M10
operation-type      Media
port                0
slot                1
virtual-mac
admin-state         enabled
auto-negotiation    enabled
duplex-mode         FULL
speed               100
overload-protection disabled
```


5. Configure Network Interface values

To configure Network Interface values, use the network-interface command under the system branch. To enter the system branch from phy-interface, you issue the **exit** command then enter the **network-interface** command.

You will first configure the IP characteristics for the M10 interface defined above. A hostname for the network-interface is defined which represents the FQDN of the PSTN gateway in Lync topology and this FQDN will be configured as common-name in the Certificate-record when configuring TLS on the E-SBC

```
ACME1A(phy-interface)# exit
ACME1A(system)# network-interface
ACME1A(network-interface)# name M10
ACME1A(network-interface)# description "Mediation Server-facing interface"
ACME1A(network-interface)# hostname acme1.st02.loc
ACME1A(network-interface)# ip-address 2.2.1.4
ACME1A(network-interface)# netmask 255.255.255.0
ACME1A(network-interface)# gateway 2.2.1.1
ACME1A(network-interface)# dns-ip-primary 2.2.1.5
ACME1A(network-interface)# dns-domain st02.loc
ACME1A(network-interface)#add-hip-ip 2.2.1.4
ACME1A(network-interface)#add-icmp-ip 2.2.1.4
ACME1A(network-interface)# done
```

```
network-interface
name M10
sub-port-id 0
description Mediation Server-facing interface
hostname acme1.st02.loc
ip-address 2.2.1.4
pri-utility-addr
sec-utility-addr
netmask 255.255.255.0
gateway 2.2.1.1
sec-gateway
gw-heartbeat
state disabled
heartbeat 0
retry-count 0
retry-timeout 1
health-score 0
dns-ip-primary 2.2.1.5
dns-ip-backup1
dns-ip-backup2
dns-domain st02.loc
dns-timeout 11
hip-ip-list 2.2.1.4
ftp-address
```

```
icmp-address          2.2.1.4
snmp-address
telnet-address
ssh-address
```

You will now configure the slot 0 port 0 subport 0 network-interface, specifying the appropriate values.

```
ACME1A(network-interface) # name M00
ACME1A(network-interface) # description "SIP trunk facing interface"
ACME1A(network-interface) # ip-address 10.10.1.4
ACME1A(network-interface) # netmask 255.255.255.0
ACME1A(network-interface) # gateway 10.10.1.1
ACME1A(network-interface) # add-hip-ip 10.10.1.4
ACME1A(network-interface) # add-icmp-ip 10.10.1.4
ACME1A(network-interface) # done
```

```
network-interface
name                M00
sub-port-id         0
description         SIP Trunk facing interface
hostname
ip-address          10.10.1.4
pri-utility-addr
sec-utility-addr
netmask             255.255.255.0
gateway             10.10.1.10
sec-gateway
gw-heartbeat
state               disabled
heartbeat           0
retry-count         0
retry-timeout       1
health-score        0
dns-ip-primary
dns-ip-backup1
dns-ip-backup2
dns-domain
dns-timeout         11
hip-ip-list         10.10.1.4
ftp-address
icmp-address        10.10.1.4
snmp-address
telnet-address
ssh-address
```

6. Configure Global SIP configuration

To configure the Global SIP values, use the **sip-config** command under the session-router branch. To enter the session-router branch from network-interface, you issue the **exit** command twice, followed by the **sip-config** command.

```
ACME1A(network-interface)# exit
ACME1A(system)# exit
ACME1A(configure)# session-router
ACME1A(session-router)# sip-config
ACME1A(sip-config)# operation-mode dialog
ACME1A(sip-config)# home-realm-id core
ACME1A(sip-config)# extra-method-stats enabled
ACME1A(sip-config)# done
```

```
sip-config
state                enabled
operation-mode      dialog
dialog-transparency enabled
home-realm-id       core
egress-realm-id
nat-mode            None
registrar-domain
registrar-host
registrar-port      0
register-service-route always
init-timer          500
max-timer           4000
trans-expire        32
invite-expire       180
inactive-dynamic-conn 32
enforcement-profile
pac-method
pac-interval        10
pac-strategy        PropDist
pac-load-weight     1
pac-session-weight  1
pac-route-weight    1
pac-callid-lifetime 600
pac-user-lifetime   3600
red-sip-port        1988
red-max-trans       10000
red-sync-start-time 5000
red-sync-comp-time  1000
add-reason-header   disabled
sip-message-len     4096
enum-sag-match      disabled
extra-method-stats  enabled
rph-feature         disabled
```

```

nsep-user-sessions-rate      0
nsep-sa-sessions-rate       0
registration-cache-limit    0
register-use-to-for-lp      disabled
refer-src-routing           disabled
add-ucid-header             disabled
proxy-sub-events
pass-gruu-contact           disabled
sag-lookup-on-redirect      disabled

```

7. Configure Global Media configuration

To configure the Media values, use the `media-manager` command under the `media-manager` branch. To enter the `media-manager` branch from `sip-config`, you issue the **exit** command twice, followed by the **media-manager** command twice.

By issuing the **select** then **done** commands at this level, you will be creating the `media-manager` element, enabling the media management functions in the Net-Net ESD with the default values.

```


ACME1A(sip-config)# exit
ACME1A(session-router)# exit
ACME1A(configure)# media-manager
ACME1A(media-manager)# media-manager
ACME1A(media-manager-config)# select
ACME1A(media-manager-config)# done

```

```

media-manager
state                enabled
latching            enabled
flow-time-limit     86400
initial-guard-timer 300
subsq-guard-timer   300
tcp-flow-time-limit 86400
tcp-initial-guard-timer 300
tcp-subsq-guard-timer 300
tcp-number-of-ports-per-flow 2
hnt-rtcp            disabled
algd-log-level      NOTICE
mbcd-log-level      NOTICE
red-flow-port       1985
red-mgcp-port       1986
red-max-trans       10000
red-sync-start-time 5000
red-sync-comp-time  1000
media-policing      enabled
max-signaling-bandwidth 10000000
max-untrusted-signaling 100

```



min-untrusted-signaling	30
app-signaling-bandwidth	0
tolerance-window	30
rtcp-rate-limit	0
trap-on-demote-to-deny	disabled
min-media-allocation	2000
min-trusted-allocation	4000
deny-allocation	64000
anonymous-sdp	disabled
arp-msg-bandwidth	32000
fragment-msg-bandwidth	0
rfc2833-timestamp	disabled
default-2833-duration	100
rfc2833-end-pkts-only-for-non-sig	enabled
translate-non-rfc2833-event	disabled
media-supervision-traps	disabled
dnalg-server-failover	disabled

8. Configure Realms configuration

To configure the realm values, use the `realm-config` command under the `media-manager` branch. To enter the `media-manager` branch from `media-manager-config`, you issue the `exit` command, followed by the `realm-config` command.

You will create two realms:

- The `core`, which represents the mediation server-facing network; and
- The `pstn`, which represents the SIP trunk facing network.

```
ACME1A(media-manager-config)# exit
ACME1A(media-manager)# realm-config
ACME1A(realm-config)# identifier core
ACME1A(realm-config)# description "Mediation Server facing"
ACME1A(realm-config)# network-interfaces s0p0:0
ACME1A(realm-config)# done
```

```
realm-config
identifier                core
description                Mediation Server-facing
addr-prefix                0.0.0.0
network-interfaces        M10:0

mm-in-realm                disabled
mm-in-network              enabled
mm-same-ip                 enabled
mm-in-system               enabled
bw-cac-non-mm              disabled
msm-release                disabled
qos-enable                 disabled
generate-UDP-checksum      disabled
max-bandwidth              0
fallback-bandwidth         0
max-priority-bandwidth     0
max-latency                0
max-jitter                 0
max-packet-loss            0
observ-window-size         0
parent-realm
dns-realm
media-policy
media-sec-policy
in-translationid
out-translationid
in-manipulationid
out-manipulationid
manipulation-string
manipulation-pattern
```

class-profile	
average-rate-limit	0
access-control-trust-level	none
invalid-signal-threshold	0
maximum-signal-threshold	0
untrusted-signal-threshold	0
nat-trust-threshold	0
deny-period	30
ext-policy-svr	
symmetric-latching	disabled
pai-strip	disabled
trunk-context	
early-media-allow	
enforcement-profile	
additional-prefixes	
restricted-latching	none
restriction-mask	32
accounting-enable	enabled
user-cac-mode	none
user-cac-bandwidth	0
user-cac-sessions	0
icmp-detect-multiplier	0
icmp-advertisement-interval	0
icmp-target-ip	
monthly-minutes	0
net-management-control	disabled
delay-media-update	disabled
refer-call-transfer	disabled
dyn-refer-term	disabled
codec-policy	
codec-manip-in-realm	disabled
constraint-name	
call-recording-server-id	
xnq-state	xnq-unknown
hairpin-id	0
stun-enable	disabled
stun-server-ip	0.0.0.0
stun-server-port	3478
stun-changed-ip	0.0.0.0
stun-changed-port	3479
match-media-profiles	
qos-constraint	
sip-profile	
sip-isup-profile	
block-rtcp	disabled
hide-egress-media-update	disabled
last-modified-by	admin@console
last-modified-date	2010-07-15 17:01:33

You will now configure the pstn realm for SIP Trunk side of the SBC, specifying the appropriate values.

```
ACME1A (realm-config) # identifier pstn
ACME1A (realm-config) # description "To Sip Trunk"
ACME1A (realm-config) # network-interfaces M00:0
ACME1A (realm-config) # done
```

```
realm-config
identifier                pstn
description               To SIP Trunk
addr-prefix              0.0.0.0
network-interfaces
                          M00:0
mm-in-realm              disabled
mm-in-network            enabled
mm-same-ip               enabled
mm-in-system             enabled
bw-cac-non-mm            disabled
msm-release              disabled
qos-enable               disabled
generate-UDP-checksum    disabled
max-bandwidth            0
fallback-bandwidth       0
max-priority-bandwidth   0
max-latency              0
max-jitter               0
max-packet-loss          0
observ-window-size       0
parent-realm
dns-realm
media-policy
media-sec-policy
in-translationid
out-translationid
in-manipulationid
out-manipulationid
manipulation-string
manipulation-pattern
class-profile
average-rate-limit       0
access-control-trust-level none
invalid-signal-threshold 0
maximum-signal-threshold 0
untrusted-signal-threshold 0
nat-trust-threshold      0
deny-period              30
ext-policy-svr
symmetric-latching       disabled
```


pai-strip	disabled
trunk-context	
early-media-allow	
enforcement-profile	
additional-prefixes	
restricted-latching	none
restriction-mask	32
accounting-enable	enabled
user-cac-mode	none
user-cac-bandwidth	0
user-cac-sessions	0
icmp-detect-multiplier	0
icmp-advertisement-interval	0
icmp-target-ip	
monthly-minutes	0
net-management-control	disabled
delay-media-update	disabled
refer-call-transfer	disabled
dyn-refer-term	disabled
codec-policy	
codec-manip-in-realm	disabled
constraint-name	
call-recording-server-id	
xnq-state	xnq-unknown
hairpin-id	0
stun-enable	disabled
stun-server-ip	0.0.0.0
stun-server-port	3478
stun-changed-ip	0.0.0.0
stun-changed-port	3479
match-media-profiles	
qos-constraint	
sip-profile	sip-profile
sip-isup-profile	
block-rtcp	disabled
hide-egress-media-update	disabled
last-modified-by	admin@console
last-modified-date	2010-07-15 17:02:11

9. Configure SIP signaling configuration

To configure the SIP signaling values, use the **sip-interface** command under the session-router branch. To enter the session-router branch from realm-config, you issue the **exit** command twice, followed by the **sip-interface** command.

Here you will be configuring the IP addresses and TCP ports on which the Net-Net SD will listen for and transmit SIP messages. These will be the same IP addresses as configured on the associated network-interface elements.

```

ACME1A(realm-config)# exit
ACME1A(media-manager)# exit
ACME1A(configure)# session-router
ACME1A(session-router)# sip-interface
ACME1A(sip-interface)# realm pstn
ACME1A(sip-interface)# description "SIP Trunk facing"
ACME1A(sip-interface)# sip-ports
ACME1A(sip-port)# address 10.10.1.4
ACME1A(sip-port)# transport-protocol TCP
ACME1A(sip-port)# allow-anonymous agents-only
ACME1A(sip-port)# done

```

```

sip-port
address                10.10.1.4
port                   5060
transport-protocol    TCP
tls-profile
allow-anonymous       agents-only
ims-aka-profile

```

To ensure that the SBC is doing topology hiding and replacing host-portions in SIP URIs of From and To headers, in-built sip manipulation ACME_NAT_TO_FROM_IP will need to be configured as the out-manipulationid on this sip-interface

```

ACME1A(sip-port)# exit
ACME1A(sip-interface)# out-manipulationid ACME_NAT_TO_FROM_IP
ACME1A(sip-interface)# done

```

```

sip-interface
state                enabled
realm-id             pstn
description          SIP Trunk-facing interface
sip-port
address              10.10.1.4
port                 5060
transport-protocol  TCP
tls-profile
allow-anonymous     agents-only
ims-aka-profile
carriers
trans-expire         0
invite-expire        0
max-redirect-contacts 0
proxy-mode
redirect-action
contact-mode         none
nat-traversal        none
nat-interval         30
tcp-nat-interval     90

```

registration-caching	disabled
min-reg-expire	300
registration-interval	3600
route-to-registrar	disabled
secured-network	disabled
teluri-scheme	disabled
uri-fqdn-domain	
trust-mode	all
max-nat-interval	3600
nat-int-increment	10
nat-test-increment	30
sip-dynamic-hnt	disabled
stop-recurse	401,407
port-map-start	0
port-map-end	0
in-manipulationid	
out-manipulationid	ACME_NAT_TO_FROM_IP
manipulation-string	
manipulation-pattern	
sip-ims-feature	disabled
operator-identifier	
anonymous-priority	none
max-incoming-conns	0
per-src-ip-max-incoming-conns	0
inactive-conn-timeout	0
untrusted-conn-timeout	0
network-id	
ext-policy-server	
default-location-string	
charging-vector-mode	pass
charging-function-address-mode	pass
ccf-address	
ecf-address	
term-tgrp-mode	none
implicit-service-route	disabled
rfc2833-payload	101
rfc2833-mode	transparent
constraint-name	
response-map	
local-response-map	
ims-aka-feature	disabled
enforcement-profile	
route-unauthorized-calls	
tcp-keepalive	none
add-sdp-invite	disabled
add-sdp-profiles	
sip-profile	
sip-isup-profile	
last-modified-by	admin@console

last-modified-date

2010-07-15 17:05:47

You will now configure the Lync mediation server-facing SIP interface.

```
ACME1A(sip-interface)# realm-id peer
ACME1A(sip-interface)# description "Mediation Server Facing interface"
ACME1A(sip-interface)# sip-ports
ACME1A(sip-port)# address 2.2.1.4
ACME1A(sip-port)# transport-protocol TCP
ACME1A(sip-port)# allow-anonymous agents-only
ACME1A(sip-port)# done
```

```
sip-port
address          2.2.1.4
port             5060
transport-protocol TCP
tls-profile
allow-anonymous  agents-only
ims-aka-profile
```

```
ACME1A(sip-port)# exit
ACME1A(sip-interface)# out-manipulationid ACME_NAT_TO_FROM_IP
ACME1AACME1A(sip-interface)# done
```

```
sip-interface
state          enabled
realm-id      peer
description   Mediation Server-facing interface
sip-port
address       2.2.1.4
port         5060
transport-protocol TCP
tls-profile
allow-anonymous agents-only
ims-aka-profile
carriers
trans-expire  0
invite-expire 0
max-redirect-contacts 0
proxy-mode
redirect-action
contact-mode  none
nat-traversal none
nat-interval  30
tcp-nat-interval 90
registration-caching disabled
min-reg-expire 300
registration-interval 3600
route-to-registrar disabled
```

```
secured-network          disabled
teluri-scheme           disabled
uri-fqdn-domain
trust-mode              all
max-nat-interval        3600
nat-int-increment       10
nat-test-increment      30
sip-dynamic-hnt         disabled
stop-recurse            401,407
port-map-start          0
port-map-end            0
in-manipulationid
out-manipulationid      ACME_NAT_TO_FROM_IP
manipulation-string
manipulation-pattern
sip-ims-feature         disabled
operator-identifier
anonymous-priority      none
max-incoming-conns      0
per-src-ip-max-incoming-conns 0
inactive-conn-timeout   0
untrusted-conn-timeout  0
network-id
ext-policy-server
default-location-string
charging-vector-mode     pass
charging-function-address-mode pass
ccf-address
ecf-address
term-tgrp-mode          none
implicit-service-route  disabled
rfc2833-payload         101
rfc2833-mode            transparent
constraint-name
response-map
local-response-map
ims-aka-feature         disabled
enforcement-profile
route-unauthorized-calls
tcp-keepalive           none
add-sdp-invite          disabled
add-sdp-profiles
sip-profile
sip-isup-profile
```

10. Configure next-hop signaling elements

To configure the next-hop signaling elements (i.e., the mediation server and service provider next-hop network element) you define session-agents. Use the **session-agent** command under the session-router branch. To enter the session-router branch from sip-interface, you issue the **exit** command, followed by the **session-agent** command.

Here you will be configuring the IP addresses and TCP ports to which the Net-Net SD will send and from which it will expect to receive SIP messages for your next-hop signaling elements.

Lync Server 2013 Gateway specification outlines the need for the SBC to have capability to do DNS load balancing among a pool of mediation servers. This is currently supported on SW version SCX6.2 by the Acme Packet SBC via DNS A or DNS SRV records, however not necessarily in a round-robin manner. In this document and testing, the SBC load balances between two mediation servers that are defined in a group (session-group) with round-robin algorithm configured. It is assumed that when using this kind of a configuration at any point another mediation server is added to the pool of servers, it will need to be explicitly configured on the SBC and added to the session-group which will be the responsibility of the enterprise network administrator.

We will first configure the service provider next-hop gateway/network element.

```
ACME1A(sip-interface)# exit
ACME1A(session-router)# session-agent
ACME1A(session-agent)# hostname 10.10.1.8
ACME1A(session-agent)# ip-address 10.10.1.8
ACME1A(session-agent)# port 5060
ACME1A(session-agent)# app-protocol sip
ACME1A(session-agent)# transport-method statictcp
ACME1A(session-agent)# realm-id pstn
ACME1A(session-agent)# done
```

```
session-agent
hostname                10.10.1.8
ip-address
port                    5060
state                   enabled
app-protocol            SIP
app-type
transport-method       StaticTCP
realm-id                pstn
egress-realm-id
description
carriers
allow-next-hop-lp      enabled
constraints             disabled
max-sessions           0
```

max-inbound-sessions	0
max-outbound-sessions	0
max-burst-rate	0
max-inbound-burst-rate	0
max-outbound-burst-rate	0
max-sustain-rate	0
max-inbound-sustain-rate	0
max-outbound-sustain-rate	0
min-seizures	5
min-asr	0
time-to-resume	0
ttr-no-response	0
in-service-period	0
burst-rate-window	0
sustain-rate-window	0
req-uri-carrier-mode	None
proxy-mode	
redirect-action	
loose-routing	enabled
send-media-session	enabled
response-map	
ping-method	
ping-interval	
ping-send-mode	keep-alive
ping-all-addresses	disabled
ping-in-service-response-codes	
out-service-response-codes	
media-profiles	
in-translationid	
out-translationid	
trust-me	disabled
request-uri-headers	
stop-recurse	
local-response-map	
ping-to-user-part	
ping-from-user-part	
li-trust-me	disabled
in-manipulationid	
out-manipulationid	
manipulation-string	
manipulation-pattern	
p-asserted-id	
trunk-group	
max-register-sustain-rate	0
early-media-allow	
invalidate-registrations	disabled
rfc2833-mode	none
rfc2833-payload	0
codec-policy	

```

enforcement-profile
refer-call-transfer          disabled
reuse-connections           NONE
tcp-keepalive               none
tcp-reconn-interval         0
max-register-burst-rate     0
register-burst-window        0
sip-profile                  sip-profile
sip-isup-profile
last-modified-by            admin@console
last-modified-date          2010-07-15 17:09:46

```

You will now define the mediation server. For the sake of simplicity, two mediation servers are defined and assigned to a group called 'Mediation'. The SBC then load balances among these mediation servers.

```

acmela(session-group)# group-name Mediation
acmela(session-group)# strategy RoundRobin
acmela(session-group)# dest OP1-0704.st02.loc
acmela(session-group)# dest + OP1-0706.st02.loc
acmela(session-group)# sag-recursion enabled
acmela(session-group) # done

```

Defining Mediation Server 1

```

acmela(session-agent)# hostname OP1-0704.st02.loc
acmela(session-agent)# ip-address 2.2.1.8
acmela(session-agent)# port 5066
acmela(session-agent)# app-protocol sip
acmela(session-agent)# transport-method staticTCP
acmela(session-agent)# realm core
acmela(session-agent)# ping-method OPTIONS
acmela(session-agent)# ping-interval 60
acmela(session-agent)# refer-call-transfer enabled
acmela(session-agent)# done

```

```

session-agent
  hostname          OP1-0704.st02.loc
  ip-address        2.2.1.8
  port              5066
  state             enabled
  app-protocol      SIP
  app-type
  transport-method  StaticTCP
  realm-id          core
  egress-realm-id
  description
  carriers

```


allow-next-hop-lp	enabled
constraints	disabled
max-sessions	0
max-inbound-sessions	0
max-outbound-sessions	0
max-burst-rate	0
max-inbound-burst-rate	0
max-outbound-burst-rate	0
max-sustain-rate	0
max-inbound-sustain-rate	0
max-outbound-sustain-rate	0
min-seizures	5
min-asr	0
time-to-resume	0
ttr-no-response	0
in-service-period	0
burst-rate-window	0
sustain-rate-window	0
req-uri-carrier-mode	None
proxy-mode	
redirect-action	
loose-routing	enabled
send-media-session	enabled
response-map	
ping-method	OPTIONS;next-hop=0
ping-interval	60
ping-send-mode	keep-alive
ping-all-addresses	enabled
ping-in-service-response-codes	
out-service-response-codes	
media-profiles	
in-translationid	
out-translationid	
trust-me	disabled
request-uri-headers	
stop-recurse	
local-response-map	
ping-to-user-part	
ping-from-user-part	
li-trust-me	disabled
in-manipulationid	
out-manipulationid	
manipulation-string	
manipulation-pattern	
p-asserted-id	
trunk-group	
max-register-sustain-rate	0
early-media-allow	
invalidate-registrations	disabled

```

rfc2833-mode          none
rfc2833-payload       0
codec-policy
enforcement-profile
refer-call-transfer   enabled
reuse-connections     NONE
tcp-keepalive         none
tcp-reconn-interval   0
max-register-burst-rate 0
register-burst-window  0

```

Defining Mediation Server 2

```

acmela(session-agent) # hostname OP1-0706.st02.loc
acmela(session-agent) # ip-address 2.2.1.9
acmela(session-agent) # port 5066
acmela(session-agent) # app-protocol sip
acmela(session-agent) # transport-method staticTCP
acmela(session-agent) # realm core
acmela(session-agent) # ping-method OPTIONS
acmela(session-agent) # ping-interval 60
acmela(session-agent) # refer-call-transfer enabled
acmela(session-agent) # done

```

```

session-agent
  hostname          OP1-0706.st02.loc
  ip-address        2.2.1.9
  port              5066
  state             enabled
  app-protocol      SIP
  app-type
  transport-method  StaticTCP
  realm-id          core
  egress-realm-id
  description
  carriers
  allow-next-hop-lp  enabled
  constraints        disabled
  max-sessions       0
  max-inbound-sessions 0
  max-outbound-sessions 0
  max-burst-rate     0
  max-inbound-burst-rate 0
  max-outbound-burst-rate 0
  max-sustain-rate   0
  max-inbound-sustain-rate 0
  max-outbound-sustain-rate 0
  min-seizures       5
  min-asr            0

```

time-to-resume	0
ttr-no-response	0
in-service-period	0
burst-rate-window	0
sustain-rate-window	0
req-uri-carrier-mode	None
proxy-mode	
redirect-action	
loose-routing	enabled
send-media-session	enabled
response-map	
ping-method	OPTIONS;next-hop=0
ping-interval	60
ping-send-mode	keep-alive
ping-all-addresses	enabled
ping-in-service-response-codes	
out-service-response-codes	
media-profiles	
in-translationid	
out-translationid	
trust-me	disabled
request-uri-headers	
stop-recurse	
local-response-map	
ping-to-user-part	
ping-from-user-part	
li-trust-me	disabled
in-manipulationid	
out-manipulationid	
manipulation-string	
manipulation-pattern	
p-asserted-id	
trunk-group	
max-register-sustain-rate	0
early-media-allow	
invalidate-registrations	disabled
rfc2833-mode	none
rfc2833-payload	0
codec-policy	
enforcement-profile	
refer-call-transfer	enabled
reuse-connections	NONE
tcp-keepalive	none
tcp-reconn-interval	0
max-register-burst-rate	0
register-burst-window	0

11. Configure SIP routing

To configure the SIP routing, use the **local-policy** command under the session-router branch. To enter the session-router branch from session-agent, you issue the **exit** command, followed by the **local-policy** command.

We will first configure the route from the gateway to the mediation server.

```
ACME1A(session-agent)# exit
ACME1A(session-router)# local-policy
ACME1A(local-policy)# from-address *
ACME1A(local-policy)# to-address *
ACME1A(local-policy)# source-realm pstn
ACME1A(local-policy)# policy-attributes
ACME1A(local-policy-attributes)# next-hop SAG:Mediaton
ACME1A(local-policy-attributes)# action replace-uri
ACME1A(local-policy-attributes)# app-protocol sip
ACME1A(local-policy-attributes)# done
```

```
policy-attribute
next-hop                SAG:Mediation
                        realm
                        action                replace-uri
                        terminate-recursion    disabled
                        carrier
                        start-time            0000
                        end-time              2400
                        days-of-week          U-S
                        cost                  0
                        app-protocol          SIP
                        state                  enabled
                        methods
                        media-profiles
                        lookup                single
                        next-key
                        eloc-str-lkup          disabled
                        eloc-str-match
```

```
ACME1A(local-policy-attributes)# exit
ACME1A(local-policy)# done
```

```
local-policy
from-address                *
to-address                  *
source-realm                pstn
description
```

```

activate-time          N/A
deactivate-time       N/A
state                 enabled
policy-priority       none
next-hop              SAG:Mediation

    realm
    action             replace-uri
    terminate-recursion disabled
    carrier
    start-time         0000
    end-time           2400
    days-of-week       U-S
    cost               0
    app-protocol       SIP
    state              enabled
    methods
    media-profiles
    lookup             single
    next-key
    eloc-str-lkup     disabled
    eloc-str-match

```

We will now configure the route from the mediation server to the gateway.

```

ACME1A(local-policy)# from-address *
ACME1A(local-policy)# to-address *
ACME1A(local-policy)# source-realm core
ACME1A(local-policy)# policy-attributes
ACME1A(local-policy-attributes)# next-hop 10.10.1.8
ACME1A(local-policy-attributes)# realm pstn
ACME1A(local-policy-attributes)# app-protocol sip
ACME1A(local-policy-attributes)# done

```

```

policy-attribute
next-hop          10.10.1.8
realm            pstn
action           replace-uri
terminate-recursion disabled
carrier
start-time       0000
end-time         2400
days-of-week    U-S
cost             0
app-protocol     SIP
state           enabled
methods
media-profiles
lookup          single
next-key

```

```
eloc-str-lkup          disabled
eloc-str-match
```

```
ACME1A(local-policy-attributes)# exit
ACME1A(local-policy)# done
```

```
local-policy
  from-address          *
  to-address            *

  source-realm
                      core

  description
  activate-time        N/A
  deactivate-time      N/A
  state                enabled
  policy-priority      none
  policy-attribute
  next-hop              10.10.1.8
  realm                pstn
  action                replace-uri
  terminate-recursion  disabled
  carrier
  start-time           0000
  end-time             2400
  days-of-week         U-S
  cost                 0
  app-protocol         SIP
  state                enabled
  methods
  media-profiles
  lookup               single
  next-key
  eloc-str-lkup        disabled
  eloc-str-match
```

11 a. Call Transfer Scenarios

Lync Server 2013 authorizes transfers of all Lync initiated calls whether it is Lync to Lync or Lync to PSTN. Acme Packet Net-Net SBC provides REFER handling by terminating the REFER from Lync and generating an INVITE for the referred party towards Lync Mediation server. Lync then process the INVITE, authorizes the call transfer and sends either a new INVITE (for call transferred to PSTN) to the SBC or transfers call internally to transferred Lync client

To handle call transfer and refer scenarios – when Lync client 1 refers/transfers the call to Lync 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

```

ACME1A(local-policy) # from-address *
ACME1A(local-policy) # to-address OP1-0704.st02.loc
ACME1A(local-policy) # source-realm pstn
ACME1A(local-policy) # description "for referred party OP1-0704.st02.loc"
ACME1A(local-policy) # policy-attributes
ACME1A(local-policy-attributes) # next-hop OP1-0704.st02.loc
ACME1A(local-policy-attributes) # realm core
ACME1A(local-policy-attributes) # action replace-uri
ACME1A(local-policy-attributes) # done
ACME1A(local-policy-attributes) # exit
ACME1A(local-policy) # done
ACME1A(local-policy) # from-address *
ACME1A(local-policy) # to-address OP1-0706.st02.loc
ACME1A(local-policy) # source-realm pstn
ACME1A(local-policy) # description "for referred party OP1-0706.st02.loc"
ACME1A(local-policy) # policy-attributes
ACME1A(local-policy-attributes) # next-hop OP1-0706.st02.loc
ACME1A(local-policy-attributes) # realm core
ACME1A(local-policy-attributes) # action replace-uri
ACME1A(local-policy-attributes) # done
ACME1A(local-policy-attributes) # exit
ACME1A(local-policy) # done

```

```

local-policy
  from-address          *
  to-address            OP1-0704.st02.loc
  source-realm          pstn
  description           for referred party OP1-
0704.st02.loc
  activate-time         N/A
  deactivate-time       N/A
  state                 enabled
  policy-priority       none
  last-modified-by      admin@10.176.33.30
  last-modified-date    2011-06-22 14:46:32
  policy-attribute
    next-hop            OP1-0704.st02.loc
    realm               core
    action               replace-uri
    terminate-recursion disabled
    carrier
    start-time          0000
    end-time             2400

```

```

days-of-week      U-S
cost               0
app-protocol      SIP
state             enabled
methods
media-profiles
lookup            single
next-key
eloc-str-lkup     disabled
eloc-str-match

local-policy
  from-address
                  *
  to-address
                  OP1-0706.st02.loc
  source-realm
                  pstn
  description
0706.st02.loc    for referred party OP1-
  activate-time   N/A
  deactivate-time N/A
  state          enabled
  policy-priority none
  last-modified-by admin@10.176.33.30
  last-modified-date 2011-06-22 14:47:35
  policy-attribute
    next-hop      OP1-0706.st02.loc
    realm         core
    action        replace-uri
    terminate-recursion disabled
    carrier
    start-time    0000
    end-time      2400
    days-of-week  U-S
    cost         0
    app-protocol  SIP
    state        enabled
    methods
    media-profiles
    lookup        single
    next-key
    eloc-str-lkup disabled
    eloc-str-match

```

12. Configure media handling

To configure the media handling, use the **steering-pool** command under the media-manager branch. To enter the steering-pool branch from local-policy, you issue the **exit** command twice, followed by the **media-manager** then the **steering-pool** command.

You will use the same IP address for the steering pool as the one used for the SIP interface. Note that the port ranges provide a means of limiting the number of concurrent media sessions within a given realm. For example, assigning 100 ports to a realm would limit it to 50 concurrent bidirectional calls, where two ports are assigned (one per unidirectional media stream).

```
ACME1A(local-policy)# exit
ACME1A(session-router)# exit
ACME1A(configure)# media-manager
ACME1A(media-manager)# steering-pool
ACME1A(steering-pool)# ip-address 10.10.1.8
ACME1A(steering-pool)# start-port 40000
ACME1A(steering-pool)# end-port 60000
ACME1A(steering-pool)# realm-id pstn
ACME1A(steering-pool)# done
```

```
steering-pool
ip-address          10.10.1.8
start-port         40000
end-port           60000
realm-id           pstn
network-interface
```

You will now configure the media handling for the core (Lync mediation server) realm.

```
ACME1A(steering-pool)# ip-address 2.2.1.4
ACME1A(steering-pool)# start-port 40000
ACME1A(steering-pool)# end-port 60000
ACME1A(steering-pool)# realm-id core
ACME1A(steering-pool)# done
```

```
steering-pool
ip-address          2.2.1.4
start-port         40000
end-port           60000
realm-id           core
network-interface
```

13. Configuring SIP PRACK interworking

In order to establish an early media session for outbound calls, Lync Server 2013 gateway specification mandates the PSTN gateways to offer a reliable provisional response and for inbound calls offer INVITEs with supported header. The SBC can interwork and provide RFC

3262 PRACK interworking towards Lync and it is mandatory configuration in all Acme Packet – Microsoft Lync deployments. For this the following need to be configured:

- 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 to add a Require:100rel header in incoming SIP INVITE from mediation server and delete the Supported:100rel header

```
ACME1A(session-router)# sip-interface
ACME1A(sip-interface)# select
<realm-id>:
1: core          2.2.1.4:5067
2: pstn          10.10.1.4:5060
selection: 1
ACME1A(sip-interface)#options 100rel-interworking
```

Configure Sip-feature to pass Supported and Require headers in SIP messages

```
ACME1A(session-router)#sip-feature
ACME1A(sip-feature)#name 100rel-interworking
ACME1A(sip-feature)#realm pstn
ACME1A(sip-feature)# support-mode-inbound Pass
ACME1A(sip-feature)# require-mode-inbound Pass
ACME1A(sip-feature)# proxy-require-mode-inbound Pass
ACME1A(sip-feature)# support-mode-outbound Pass
ACME1A(sip-feature)# require-mode-outbound Pass
ACME1A(sip-feature)# proxy-require-mode-outbound Pass
ACME1A(sip-feature)#done
```

```
sip-feature
  name          100rel
  realm         pstn
  support-mode-inbound  Pass
  require-mode-inbound  Pass
  proxy-require-mode-inbound  Pass
  support-mode-outbound  Pass
  require-mode-outbound  Pass
  proxy-require-mode-outbound  Pass
```

```
ACME1A(sip-manipulation)# name Forearlymedia
ACME1A(sip-manipulation)# header-rules
ACME1A(sip-header-rules)# name delsupported
ACME1A(sip-header-rules)# header-name Supported
ACME1A(sip-header-rules)# action delete
ACME1A(sip-header-rules)# comparison-type case-sensitive
ACME1A(sip-header-rules)# msg-type request
ACME1A(sip-header-rules)# methods INVITE
ACME1A(sip-header-rules)# done
ACME1A(sip-header-rules)# name addrequireinINVITE
ACME1A(sip-header-rules)# header-name Require
ACME1A(sip-header-rules)# action add
ACME1A(sip-header-rules)# comparison-type case-sensitive
```

```

ACME1A(sip-header-rules)# msg-type request
ACME1A(sip-header-rules)# methods INVITE
ACME1A(sip-header-rules)# done
ACME1A(sip-header-rules)# exit
ACME1A(sip-manipulation)# done

```

```

sip-manipulation
  name                               Forearlymedia
  description
  split-headers
  join-headers
  header-rule
    name                               delsupported
    header-name                         Supported
    action                               delete
    comparison-type                     case-sensitive
    msg-type                             request
    methods                              INVITE
    match-value
    new-value
  header-rule
    name                               addrequireinINVITE
    header-name                         Require
    action                               add
    comparison-type                     case-sensitive
    msg-type                             request
    methods                              INVITE

```

Reference the sip-manipulation name/id as an in-manipulationid on the 2.2.1.4 sip-interface

```

ACME1A(session-router)# sip-interface
ACME1A(sip-interface)# select
<realm-id>:
1: core          2.2.1.4:5067
2: pstn         10.10.1.4:5060
selection: 1
ACME1A(sip-interface)#in-manipulationid Forearlymedia
ACME1A(sip-interface)#done

```

14. TLS & SRTP configuration on Net-Net SD

In some applications, it may be required for the E-SBC to establish a TLS connection and use media encryption (support SRTP) with Lync server. The Net-Net ESD can interwork and terminate SIP over TLS and SRTP between Lync and the Sip trunk provider. This portion of the guide provides instructions on how to achieve this. Note that the Net-Net ESD must have the appropriate hardware (IPSec NIUs) and at a minimum a software TLS license to support this capability. If you have any questions regarding these, please contact your Acme Packet representative.

14.1 TLS Configuration on the Net-Net SD

To configure TLS on the Net-Net Session Director, the following steps will need to be followed:

- Create certificate-record, generate certificate request and import signed certificate
- Create a TLS profile
- Apply tls-profile on the sip-interface
- Change transport-protocol on Session-agent (from TCP to TLS)

14 1.1 Create certificate-record

To configure certificate-record use the certificate-record menu under the security branch. Since Lync server requires mutual TLS authentication, both parties (SBC and mediation server) will need to provide their own certificate to the peer during the TLS handshake for the purpose of authenticating themselves to each other.

Create certificate record holder for Microsoft Certificate authority (Enterprise CA)

```
Acmela(configure terminal)#security
Acmela(security)#certificate-record
Acmela(certificate-record)#name ST02-OP1-0703-CA
Acmela(certificate-record)#country US
Acmela(certificate-record)#state WA
Acmela(certificate-record)locality Redmond
Acmela(certificate-record)key-size 2048
Acmela(certificate-record)done
```

```
certificate-record
  name                ST02-OP1-0703-CA
  country              US
  state                WA
  locality             Redmond
  organization         loc
  unit                 ST02
  common-name
  key-size             2048
  alternate-name
  trusted              enabled
  key-usage-list
                     digitalSignature
                     keyEncipherment
  extended-key-usage-list
                     serverAuth
```

You will now configure an additional certificate record holder for the Net-Net Session Director (end-entity certificate). The common-name needs to be an FQDN per Lync server 2013 gateway specification and is exchanged in the CN part of the Subject field of X.509 TLS certificate that is presented by the Net-Net SBC. This FQDN is also provisioned in Lync that resolves to the Lync mediation server facing Sip-interface IP address of the SBC

```
Acmela(certificate-record)#name acme-rcrd
Acmela(certificate-record)#country US
Acmela(certificate-record)# state WA
Acmela(certificate-record)#common-name acme1.st02.loc
Acmela(certificate-record)#key-size 2048
Acmela(certificate-record)#done
```

```
certificate-record
  name                acme-rcrd
  country              US
  state                WA
  locality             Redmond
  organization         loc
  unit                 ST02
  common-name          acmesbc.acmepacket.net
  key-size             2048
  alternate-name
  trusted              enabled
  key-usage-list
                     digitalSignature
                     keyEncipherment
  extended-key-usage-list
                     serverAuth
```

Once the certificate records are created, you will generate a certificate request and have the CA sign the SBC certificate request

As an example a screenshot below is provided to generate a certificate request and import the signed certificate from the CA

```
comptnr-ddos# generate-certificate-request acme-rcrd
Generating Certificate Signing Request. This can take several minutes....
```

```
-----BEGIN CERTIFICATE REQUEST-----
MIIB1jCCAT8CAQAwwDELMAkGA1UEBhMCVVMxCzAJBgNVBAGTAk1BMRMwEQYDVQQH
EwpCdXJsaW5ndG9uMRQwEgYDVQQKEwtFbmdpbmVlcmluZzENMAsGA1UEAxMEYWnt
ZTCBnzANBgkqhkiG9w0BAQEFAAOBjQAwgYkCgYEAwJF6D8gnSvc9Ug4aCkyIqQeW
kgkyk+kNYXESQNoOHJTdre9R6IB5f1UdsM/8klTTpyBwMn+GooeOc8iJ4wks+7VG
QegsZEawyDFoTC1wQCpPwSEFjOQWtnybp3ZLVBBPJ4Mcowi7qKJK/Pe4aCgfEhue
27BwC/HHD2lZIMlyhyECAwEAAaBCMBQGA1UdJTENEwtzZXJ2ZXJBdXRoIDAgBgNV
HQ8xIxBhZGlnaXRhbFNPZ25hdHVyZSBrZXI1FbMnpeGhlcm1lbnQgMA0GCSqGSIb3
DQEBBQUAA4GBAK0wXNVG33sBrNcLCndMrzEA5xONY7q2f2PXhm7dfgrgS2e2XNoZ
rqeh127aBTcgCidRAEGYVL1EVFOCT1Mo9++B7dyfa3K5eG6m78GILqoonZluIqba
Y7Z0kFnAgIyR65ZjpmjFzlhNjrNtH4qWyt49fDWC7NfKrNKd8o1A16U9
-----END CERTIFICATE REQUEST-----
```

WARNING: Configuration changed, run "save-config" command.

Once the certificate request is generated, be sure to run the save-config and activate-config command. The certificate request can be FTP'd to the CA (the highlighted portion is saved in a text file)

FTP the signed certificate file "xyz.crt" to /ramdrv/; or copy to clipboard so that you can paste from ACLI (example shown below)

```
comptnr-ddos# import-certificate try-all acme-rcrd
IMPORTANT:
    Please enter the certificate in the PEM format.
    Terminate the certificate with ";" to exit.....
-----BEGIN CERTIFICATE-----
MIICQDCCAakCAQQwDQYJKoZIhvcNAQEFBQAwfTELMakGA1UEBhMCMVVMxCzAJBgNV
BAGTAK1BMRMwEQYDVQQHEwPCdXJsaW5ndG9uMQ0wCwYDVQQKEwRBY211MQswCQYD
VQQLEwJlZTEOMAwGA1UEAxMFU21tb24xIDAeBgkqhkiG9wOBCQEWEXNsZWVsc2NA
eWFob28uY29tMB4XDTA5MDUxMzIwMzExOVowXDTEwMDUxMzIwMzExOVowVDELMAkG
A1UEBhMCMVVMxCzAJBgNVBAGTAK1BMRMwEQYDVQQHEwPCdXJsaW5ndG9uMRQwEgYD
VQQKEwtFbmdpbmVlcmluZzENMA5GA1UEAxMEYWNtZTCBnzANBgkqhkiG9wOBAQEF
AAOBjQAwgYkCgYEAvJF6D8gnSvc9Ug4aCkyIqQeWkgkyk+kNYXESQNoOHJTdre9R
6IB5f1UdsM/8klTTpyBwMn+GoocOc8iJ4wks+7VGQegsZEawyDFoTC1wQCpUwEF
jOQWtnybp3ZLVBBPJ4Mcowi7qKJK/Pe4aCgfEhue27BwC/HHD21ZIMlyhyECAwEA
ATANBgkqhkiG9wOBAQUFAAOBgQCByeJQ/35H3FtCKtGivKQ19jOunHCynUQHU/eO
DVzUswBlzW+MpOCiz/2fo4eFYNFUrKEiPsOeYSjocLkgAZMUI5n/x3JcjQX6EiRu
8doByx8DQoEoSiqEbVOBa7fQoZMTke6YMjpnIatEg9Z5seV1AZjgMTTh/p+O3r+7
1jlmA==
-----END CERTIFICATE-----;

Certificate imported successfully....
WARNING: Configuration changed run "save-config" command
```

When you see the "Certificate imported successfully" message, ensure the save-config and activate-config commands are run. As a tip, look out for terminal client (like PuTty, Tera Term, etc.) related signed certificate text copy/paste error issues, which may or may not be successful and you may run into certificate import errors. You can delete certificate-record configuration objects and issue a save-config/activate-config to start again (in case of issues) and remove the private key in the SBC associated with the previous certificate

14.1.2 Create a tls-profile

To create a TLS profile use the tls-profile menu under the security branch.

```
Acmela(configure terminal)# security
Acmela(security)#tls-profile
Acmela(tls-profile)#name core
Acmela(tls-profile)#end-entity-certificate acmelaServerCert
Acmela(tls-profile)# trusted-ca-certificates ST02-OP1-0703-CA
Acmela(tls-profile)#done
```

```

tls-profile
  name                core
  end-entity-certificate acmelaServerCert
  trusted-ca-certificates
                        ST02-OP1-0703-CA

  cipher-list

  verify-depth        ALL
  mutual-authenticate enabled
  tls-version          compatibility
  cert-status-check   disabled
  cert-status-profile-list

```

14.1.3 Apply TLS profile on the sip-interface

Exit out of the `tls-profile` sub-menu and security branch and enter `session-router`, `sip-interface` sub-menu.

```

Acmela(tls-profile)#exit
Acmela(security)#exit
Acmela(configure)#session-router
Acmela(session-router)#sip-interface
Acmela(sip-interface)#select
<realm-id>:
1: core      2.2.1.4:5067
2: pstn     10.10.1.4:5060
selection: 1
Acmela(sip-interface)#sip-ports
Acmela(sip-port)#address 2.2.1.4
Acmela(sip-port)#port 5067
Acmela(sip-port)#tls-profile core
Acmela(sip-port)#transport-protocol TLS
Acmela(sip-port)#allow-anonymous agents-only
Acmela(sip-port)#done

```

```

sip-port
  address      2.2.1.4
  port         5067
  transport-protocol TLS
  tls-profile  core
  allow-anonymous agents-only
  ims-aka-profile
Acmela(sip-port)#exit
Acmela(sip-interface)#done

```

```

sip-interface
  state          enabled
  realm-id       core
  description
  sip-port
    address      2.2.1.4
    port         5067
    transport-protocol TLS

```

tls-profile	core
allow-anonymous	agents-only
ims-aka-profile	
carriers	
trans-expire	0
invite-expire	0
max-redirect-contacts	0
proxy-mode	
redirect-action	
contact-mode	none
nat-traversal	none
nat-interval	30
tcp-nat-interval	90
registration-caching	disabled
min-reg-expire	300
registration-interval	3600
route-to-registrar	disabled
secured-network	disabled
teluri-scheme	disabled
uri-fqdn-domain	
options	100rel-interworking
trust-mode	all
max-nat-interval	3600
nat-int-increment	10
nat-test-increment	30
sip-dynamic-hnt	disabled
stop-recurse	401,407
port-map-start	0
port-map-end	0
in-manipulationid	Forearlymedia
out-manipulationid	
manipulation-string	
manipulation-pattern	
sip-ims-feature	disabled
operator-identifier	
anonymous-priority	none
max-incoming-conns	0
per-src-ip-max-incoming-conns	0
inactive-conn-timeout	0
untrusted-conn-timeout	0
network-id	
ext-policy-server	
default-location-string	
charging-vector-mode	pass
charging-function-address-mode	pass
ccf-address	
ecf-address	
term-tgrp-mode	none
implicit-service-route	disabled
rfc2833-payload	101
rfc2833-mode	transparent
constraint-name	
response-map	
local-response-map	
ims-aka-feature	disabled
enforcement-profile	
route-unauthorized-calls	


```

tcp-keepalive           none
add-sdp-invite          disabled
add-sdp-profiles
sip-profile
sip-isup-profile

```

14.2. SRTP Configuration on Net-Net SD

Configuration elements – A brief explanation of the elements needed for SRTP configuration is provided below:

- Configure sdes (or mikey) profile to define the algorithm and cryptos to be used
- Configure a media-sec-policy to instruct the SBC on how to handle media related parameters in SDP received/sent in a realm (RTP, SRTP). Media-sec-policy will be referenced in the realm
- Configure security-policy element which creates a security association in the SBC to do the SRTP encryption and decryption

14.2.1 Algorithm and Crypto configuration on the SBC

Firstly you would configure an element which defines the algorithm and cryptos to be used which is the sdes or mikey profile. Exit out of the sip-interface/session-router branch and go to security -- >media-security -- >sdes-profile

```

Acmela(sdes-profile)#name sdes1
Acmela(sdes-profile)#crypto-list "AES_CM_128_HMAC_SHA1_80
AES_CM_128_HMAC_SHA1_32"
Acmela(sdes-profile)#egress-offer-format same-as-ingress
Acmela(sdes-profile)#done

```

```

sdes-profile
  name                sdes1
  crypto-list          AES_CM_128_HMAC_SHA1_80
AES_CM_128_HMAC_SHA1_32
  srtp-auth            enabled
  srtp-encrypt         enabled
  srtcp-encrypt        enabled
  mki                  disabled
  egress-offer-format  same-as-ingress
  use-ingress-session-params
  key
  salt

```

14.2.2 Media Security Policy for SRTP and RTP

Configure a media-sec-policy for SRTP and RTP and reference it in the appropriate realms (srtp policy for mediation server facing realm and rtp for SIP Trunk facing realm). Exit out of the sdes-profile sub-menu and go to media-sec-policy under security branch

```
Acmela(security)#media-sec-policy
Acmela(media-sec-policy)name sdespolicy
Acmela(media-sec-policy)inbound
Acmela(media-sec-inbound)#profile sdes1
Acmela(media-sec-inbound)#mode srtp
Acmela(media-sec-inbound)#protocol sdes
Acmela(media-sec-inbound)#done
```

```
inbound
      profile                sdes1
      mode                   srtp
      protocol               sdes
Acmela(media-sec-inbound)#
```

```
Acmela(media-sec-inbound)#exit
Acmela(media-sec-policy)#outbound
Acmela(media-sec-outbound)# profile sdes1
Acmela(media-sec-outbound)#mode srtp
Acmela(media-sec-outbound)#protocol sdes
Acmela(media-sec-outbound)#done
outbound
      profile                sdes1
      mode                   srtp
      protocol               sdes
Acmela(media-sec-outbound)#exit
Acmela(media-sec-policy)#done
```

```
media-sec-policy
  name                sdespolicy
  pass-through        disabled
  inbound
    profile           sdes1
    mode              srtp
    protocol          sdes
  outbound
    profile           sdes1
    mode              srtp
    protocol          sdes
  last-modified-by    admin@10.80.20.43
  last-modified-date  2011-04-28 16:55:45
```

```
Acmela(media-sec-policy)#name rtponly
Acmela(media-sec-policy)#inbound
Acmela(media-sec-inbound)#mode rtp
Acmela(media-sec-policy)#done
```



```
inbound
      profile
      mode                rtp
      protocol            none
Acmela (media-sec-inbound) #
```

```
Acmela (media-sec-policy) #outbound
Acmela (media-sec-outbound) # mode rtp
Acmela (media-sec-outbound) #done
outbound
      profile
      mode                rtp
      protocol            none
Acmela (media-sec-outbound) #
Acmela (media-sec-outbound) #exit
Acmela (media-sec-policy) #done
```

```
media-sec-policy
  name                rtponly
  pass-through        disabled
  inbound
    profile
    mode                rtp
    protocol            none
  outbound
    profile
    mode                rtp
    protocol            none
  last-modified-by    admin@10.176.33.21
  last-modified-date  2011-03-29 17:17:43
```

Once media-sec-policy is configured, it will need to be referenced in the realms. Select core realm from the realm-config sub-menu and reference media-sec-policy sdespolicy. Select pstn realm and reference the rtponly policy.

14.2.3. Configure Security Policy

Configure security-policy to create security association in the SBC for encryption/decryption of SRTP. Since the same network-interface will be associated with doing SRTP, you will need to define an explicit tls signal and dns security policy with action set to allow to permit the interface to allow tls sip signaling and dns queries. Exit out of the realm-config/media-manager branch and go to security/ipsec/security-policy

```
Acmela(security-policy)#name core-tls-signal
Acmela(security-policy)#network-interface M10:0
Acmela(security-policy)#priority 1
Acmela(security-policy)# local-ip-addr-match 2.2.1.4
Acmela(security-policy)# local-port-match 5067
Acmela(security-policy)#action allow
Acmela(security-policy)#done
```

```
security-policy
  name core-tls-signal
  network-interface M10:0
  priority 1
  local-ip-addr-match 2.2.1.4
  remote-ip-addr-match 0.0.0.0
  local-port-match 5067
  remote-port-match 0
  trans-protocol-match ALL
  direction both
  local-ip-mask 255.255.255.255
  remote-ip-mask 0.0.0.0
  action allow
  ike-sainfo-name
  outbound-sa-fine-grained-mask
    local-ip-mask 255.255.255.255
    remote-ip-mask 255.255.255.255
    local-port-mask 0
    remote-port-mask 0
    trans-protocol-mask 0
    valid enabled
    vlan-mask 0xFFF
  last-modified-by admin@10.176.33.21
  last-modified-date 2011-03-29 15:43:58
```

```
Acmela(security-policy)#name core-srtp
Acmela(security-policy)#network-interface M10:0
Acmela(security-policy)#priority 100
Acmela(security-policy)# local-ip-addr-match 2.2.1.4
Acmela(security-policy)#action srtp
Acmela(security-policy)# outbound-sa-fine-grained-mask
Acmela(outbound-sa-fine-grained-mask)#select
Acmela(outbound-sa-fine-grained-mask)#local-ip-mask 0.0.0.0
Acmela(outbound-sa-fine-grained-mask)#remote-port-mask 65535
Acmela(outbound-sa-fine-grained-mask)#trans-protocol-mask 255
Acmela(outbound-sa-fine-grained-mask)#done
```

```
outbound-sa-fine-grained-mask
    local-ip-mask          0.0.0.0
    remote-ip-mask         255.255.255.255
    local-port-mask        0
    remote-port-mask       65535
    trans-protocol-mask    255
    valid                  enabled
    vlan-mask              0xFFF
```

```
Acmela (outbound-sa-fine-grained-mask)#
Acmela (outbound-sa-fine-grained-mask)#exit
Acmela (security)#done
```

```
security-policy
    name                   core-srtp
    network-interface      M10:0
    priority               100
    local-ip-addr-match    2.2.1.4
    remote-ip-addr-match   0.0.0.0
    local-port-match       0
    remote-port-match      0
    trans-protocol-match   UDP
    direction              both
    local-ip-mask          255.255.255.255
    remote-ip-mask         0.0.0.0
    action                 srtp
    ike-sainfo-name
    outbound-sa-fine-grained-mask
        local-ip-mask      0.0.0.0
        remote-ip-mask     255.255.255.255
        local-port-mask    0
        remote-port-mask   65535
        trans-protocol-mask 255
        valid              enabled
        vlan-mask          0xFFF
    last-modified-by      admin@10.80.20.43
    last-modified-date    2011-04-28 13:15:28
```

```
Acmela(security-policy)#name core-dns
Acmela(security-policy)#network-interface M10:0
Acmela(security-policy)#priority 2
Acmela(security-policy)# local-ip-addr-match 2.2.1.4
Acmela(security-policy)# remote-ip-addr-match 2.2.1.5
Acmela(security-policy)#action allow
Acmela(security-policy)#done
```

```

security-policy
  name                core-dns
  network-interface    M10:0
  priority             2
  local-ip-addr-match 2.2.1.4
  remote-ip-addr-match 2.2.1.5
  local-port-match     0
  remote-port-match    0
  trans-protocol-match ALL
  direction            both
  local-ip-mask         255.255.255.255
  remote-ip-mask        255.255.255.255
  action               allow
  ike-sainfo-name
  outbound-sa-fine-grained-mask
    local-ip-mask      255.255.255.255
    remote-ip-mask      255.255.255.255
    local-port-mask     0
    remote-port-mask    0
    trans-protocol-mask 0
    valid               enabled
    vlan-mask           0xFFF
  last-modified-by     admin@10.80.20.43
  last-modified-date   2011-04-27 18:52:23

```

14.2.4 Change transport-method and port on Lync mediation server Session-agent

Mediation server listens for TLS signaling on port 5067 (as default) so that along with transport-method will need to be reflected in the E-SBC session-agent. Exit out from security branch and go to session-router, session-agent branch of the CLI

```

ACME1A(session-router)# session-agent
ACME1A(session-router)# select
<hostname>:

1: OP1-0704.st02.loc realm=core ip=2.2.1.8
2: OP1-0706.st02.loc realm=core ip=2.2.1.9
3: 10.10.1.8 realm=pstn ip=

Selection: 1
ACME1A(session-agent)#transport-method staticTLS
ACME1A(session-agent)#port 5067
ACME1A(session-agent)#done

```

Similarly, make the change for Mediation server 2 (with hostname OP1-0706.st02.loc)

15. Media Bypass handling

In order for Media Bypass to work, both Client and gateway (SBC) need to use the same RTP format, either SRTP (by default) or RTP. In default configuration of MS Lync, Lync client is required to use media encryption, so Media Bypass is mainly when media is encrypted (SRTP) and exchanged between Lync client and PSTN gateway (Net-Net SD).

Media Bypass from SD's perspective is to be able route RTP traffic to an endpoint/lync client on a private routable network directly (instead of RTP going to mediation server). To enable the SBC to handle media bypass feature in Lync , one will need to set restricted-latching to sdp in the core realm (facing mediation server). Select the core realm from the media-manager --- > realm-config configuration branch

```
acmela (realm-config) #restricted-latching sdp
acmela (realm-config) #done
```

```
realm-config
identifier                core
description                Mediation Server-facing
addr-prefix                0.0.0.0
network-interfaces        M10:0

mm-in-realm                disabled
mm-in-network              enabled
mm-same-ip                 enabled
mm-in-system               enabled
bw-cac-non-mm              disabled
msm-release                disabled
qos-enable                 disabled
generate-UDP-checksum      disabled
max-bandwidth              0
fallback-bandwidth         0
max-priority-bandwidth     0
max-latency                0
max-jitter                 0
max-packet-loss            0
observ-window-size         0
parent-realm
dns-realm
media-policy
media-sec-policy
in-translationid
out-translationid
in-manipulationid
out-manipulationid
manipulation-string
manipulation-pattern
class-profile
```

average-rate-limit	0
access-control-trust-level	none
invalid-signal-threshold	0
maximum-signal-threshold	0
untrusted-signal-threshold	0
nat-trust-threshold	0
deny-period	30
ext-policy-svr	
symmetric-latching	disabled
pai-strip	disabled
trunk-context	
early-media-allow	
enforcement-profile	
additional-prefixes	
restricted-latching	sdp
restriction-mask	32
accounting-enable	enabled
user-cac-mode	none
user-cac-bandwidth	0
user-cac-sessions	0
icmp-detect-multiplier	0
icmp-advertisement-interval	0
icmp-target-ip	
monthly-minutes	0
net-management-control	disabled
delay-media-update	disabled
refer-call-transfer	disabled
dyn-refer-term	disabled
codec-policy	
codec-manip-in-realm	disabled
constraint-name	
call-recording-server-id	
xnq-state	xnq-unknown
hairpin-id	0
stun-enable	disabled
stun-server-ip	0.0.0.0
stun-server-port	3478
stun-changed-ip	0.0.0.0
stun-changed-port	3479
match-media-profiles	
qos-constraint	
sip-profile	
sip-isup-profile	
block-rtcp	disabled
hide-egress-media-update	disabled

Exit out and do a save-config and an activate-config to make the configuration complete and persistent. This will make it persistent through reboots.


```
ACME1A# save-config
checking configuration
Save-Config received, processing.
waiting 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'.
ACME1A# activate-config
Activate-Config received, processing.
waiting for request to finish
Setting phy0 on Slot=0, Port=0, MAC=00:08:25:03:FC:43,
VMAC=00:08:25:03:FC:43
Setting phy1 on Slot=1, Port=0, MAC=00:08:25:03:FC:45,
VMAC=00:08:25:03:FC:45
Request to 'ACTIVATE-CONFIG' has Finished,
Activate Complete
```

16. Verify configuration integrity

You will verify your configuration referential integrity before saving and activating it with the **verify-config** command. This command is available from Superuser Mode. To enter the Superuser Mode from steering-pool, you issue the **exit** command three times.

```
ACME1A(steering-pool)# exit
ACME1A(media-manager)# exit
ACME1A(configure)# exit
ACME1A# verify-config
-----
Verification successful! No errors nor warnings in the configuration
```

Configuration is now complete.

A basic configuration on the Net-Net ESD to route calls to and from the Lync Server 2013 environment is now complete. You can now test connectivity and verify calls working following the tests outlined in the next section.

Phase III – Test connectivity to SIP Trunk

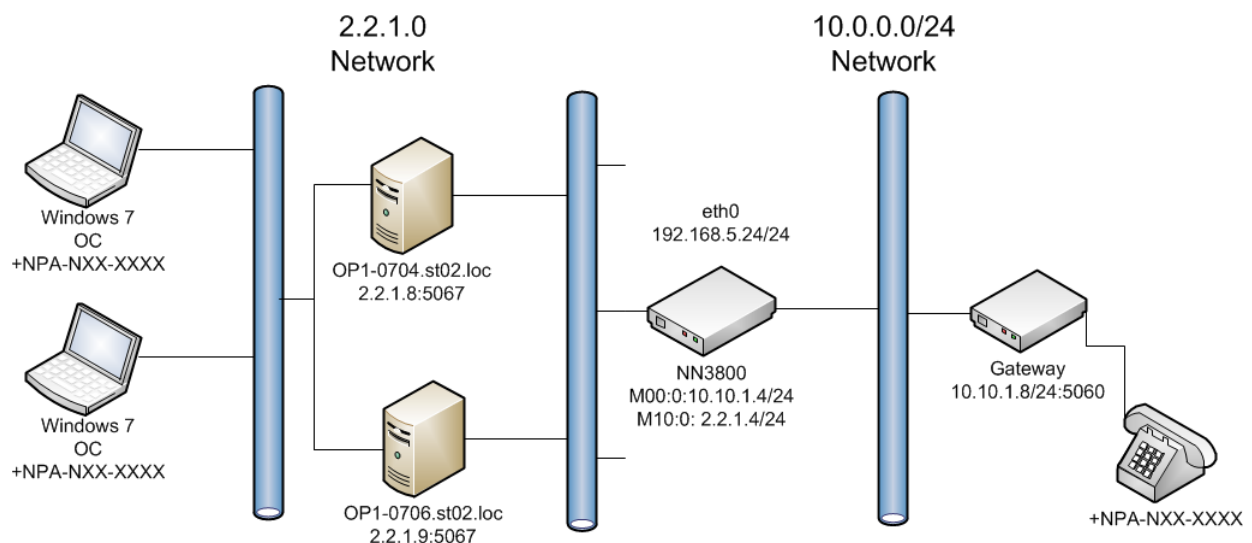
Overview

Once the Lync Server 2013 and the Net-Net Session Director have been configured, the final phase is to test connectivity and the SIP trunk interface. Acme Packet Net-Net Session Director 3820 and NN4500 have been qualified with Lync Server 2013 as part of the Unified Communications open Interoperability Program (UCOIP) test plan. Some of the test cases to verify connectivity and inbound/outbound calling are enlisted below. This section provides an overview of the topology/setup and a list of tests to verify is the deployment was successful. It is highly recommended that you use this test plan as a baseline in addition to any other tests that you may plan to run.

UCOIP Test Plan & Results

The following diagram shows the test topology.

Lync Server 2010 Acme Packet Test Topology





Test Case#	Description of Test Case	Result	Comments
408347	Lync End Point receives a call from PSTN End Point with G.711 A-law and/or G.711 U-law codecs	Pass	
408351	PSTN End Point places a call from Lync End Point on hold for 15 minutes and then resumes	Pass	
408348	PSTN End Point1 calls Lync End Point that forwards the call to PSTN End Point2	Pass	
408352	PSTN End Point calls Lync End Point1 that performs Blind Transfer to Lync End Point2 with REFER	Pass	
408349	PSTN End Point1 calls Lync End Point that escalates the call to a conference by inviting PSTN End Point2	Pass	
408350	Device fails over incoming call to Mediation Server2 when Mediation Server1 sends 503 Service Unavailable response	Pass	
408282	Device utilizes 'pool' certificates for a secure call	Pass	
408127	Device adds at least one "crypto" attribute for each media description line in the SDP	Pass	
408117	Device handles 488 Not Acceptable Here response from the Mediation Server operating in RTP only mode	Pass	
408124	Device sends Crypto attributes in SDP for call from PSTN End Point to Lync End Point	Pass	
408118	Device sends its own FQDN in contact header for TLS call from Lync End Point to PSTN End Point	Pass	
408070	PSTN End Point calls Lync End Point and hangs up while Lync End Point is still ringing	Pass	
408125	PSTN End Point calls Lync End Point with security enabled and Lync End Point later hangs up	Pass	
408080	Inbound call to Lync End Point from PSTN End Point with a very long Request-URI in the INVITE	Pass	
408065	Device correctly handles non-E.164 number in outbound Request	Pass	



	URI		
408085	Device establishes call to Lync End Point with configured value of ptime	Pass	
408063	Device generates 603 Decline response for a call rejected by PSTN End Point	Pass	
408078	Device handles call from Mediation Server with an alias name in the FROM header	Pass	
408073	Device processes call from Lync End Point with E.164 number in FROM Header URI	Pass	
408071	Device processes phone-context in Request and To URI from Lync End Point	Pass	
408066	Lync End Point calls PSTN End Point and hangs up before receiving 200 OK from Device	Pass	
408062	Lync End Point calls PSTN End Point with a call duration longer than 32 seconds	Pass	
408077	Lync End Point calls an IVR number and navigates through the IVR menu after call connection.	Pass	
408074	Lync End Point response to PSTN End Point is delayed due to network delay	Pass	
408072	Lync End Point sends INVITE with E.164 number and extension in Request and To URI	Pass	
408081	Mediation Server renegotiates an existing voice session with a different IP address	Pass	
408069	PSTN End Point disconnects established call from Lync End Point	Pass	
408068	PSTN End Point disconnects established call to Lync End Point Lync End Point disconnects established call to Lync End Point	Pass	
408067	PSTN End Point displays Lync End Point Caller ID for Outbound Call	Pass	
408183	Device negotiates Comfort Noise in a call from Lync End Point to PSTN End Point. (IPv6)	Pass	



408101	Device offers DTMF payload type in the range of 96-127 to Mediation Server	Pass	
408092	Lync End Point is able to establish a call with PSTN End Point using G.711 A-law codec	Pass	
408086	Lync End Point makes a call to PSTN End Point with G.711 A-law and/or G.711 U-law codecs	Pass	
408114	Lync End Point makes a call to PSTN End Point with G.711 U-law codec	Pass	
408119	Lync End Point receives a call from PSTN End Point with G.711 U-law codecs	Pass	
408121	Mediation Server renegotiates an existing voice session with a different RTP port	Pass	
408090	PSTN End Point is able to establish a call with Lync End Point using G.711 A-law codec	Pass	
408112	Device sends PRACK for reliable Early Media for a call from PSTN End Point to Lync End Point	Pass	
408113	Device sends PRACK for reliable Early Media for call from PSTN End Point to Lync End Point with SRTP Optional	Pass	
408064	Lync End Point calls IVR number and navigates through the IVR menu before call Connection	Pass	
408106	Lync End Point hears Early Media for a call to PSTN End Point	Pass	
408104	Device does not change the SSRC of an established inbound RTP session	Pass	
408126	Device does not change the SSRC of an established inbound SRTP session	Pass	
408100	Device does not change the SSRC of an established outbound RTP session	Pass	
408123	Device does not change the SSRC of an established outbound SRTP session	Pass	
408093	Device handles multiple RTP streams for a call to Lync End Point	Pass	



408097	Device sends RTCP packets when Lync End Point places call on hold	Pass	
408128	Device sends RTCP packets while playing music on hold	Pass	
408103	Device sends RTCP sender and receiver reports	Pass	
408105	Device sends RTCP sender and receiver reports for a secure call	Pass	
408109	Device disconnects a forked call if PSTN End Point hangs up while phones are ringing	Pass	
408079	PSTN End Point1 calls Lync End Point that is set to simultaneous ring to Lync End Point and PSTN End Point2 answers	Pass	
408110	Device disconnects a forked secure call if PSTN End Point hangs up while phones are ringing	Pass	
408094	Device handles multiple SRTP streams for a secure call to Lync End Point	Pass	
408095	Device handles multiple SRTP streams for a secure call to Lync End Point when Media Bypass is OFF	Pass	
408107	Lync End Point hears Early Media for a secure call to PSTN End Point	Pass	
408108	Lync End Point hears Early Media for a secure call to PSTN End Point when Media Bypass OFF	Pass	
408129	Lync End Point makes a secure call to PSTN End Point	Pass	
408122	Lync End Point makes a secure call to PSTN End Point and PSTN End Point later hangs up	Pass	
408130	Lync End Point makes a secure call to PSTN End Point with call duration more than 32 seconds and SRTP set to Optional	Pass	
408120	Lync End Point receives a secure call with G.711 U-law codec with Media Bypass OFF	Pass	
408091	PSTN End Point is able to establish a secure call with Lync End Point using G.711 A-law codec	Pass	
408199	Device receives History-Info headers in the SIP Invite without	Pass	



	adverse effect		
408200	3rd Party Presence headers do not cause Device failure	Pass	
408225	PSTN End Point places a call with Media Bypass OFF from Lync End Point on hold for 15 minutes and then resumes	Pass	
408235	PSTN End Point places a secure call from Lync End Point on hold and then resumes	Pass	
408224	PSTN End Point places a secure call to Lync End Point on hold and resumes after 15 minutes	Pass	
408236	PSTN End Point places a secure call to Lync End Point on hold and then resumes	Pass	
408226	PSTN End Point puts Lync End Point on hold and resumes after 15 minutes for a secure call	Pass	
408234	Lync End Point places a call from PSTN End Point on hold for 15 minutes and then resumes	Pass	
408229	Lync End Point places a call to PSTN End Point on hold and resumes after 12 minutes	Pass	
408232	Lync End Point places a secure call from PSTN End Point on hold and resumes after 15 minutes	Pass	
408230	Lync End Point places a secure call to PSTN End Point on hold and resumes after 12 minutes	Pass	
408227	Lync End Point resumes call to PSTN End Point after playing music on hold for 15 minutes	Pass	
408228	Lync End Point plays music on hold when it holds a secure call from PSTN End Point to Lync End Point	Pass	
408231	Lync End Point plays music when it holds call from PSTN End Point to Lync End Point	Pass	
408207	PSTN End Point1 calls Lync End Point that forwards all calls to PSTN End Point2 when Media Bypass OFF	Pass	
408206	PSTN End Point1 makes a secure call to Lync End Point that forwards the call to PSTN End Point2 with Media Bypass OFF	Pass	



408205	PSTN End Point1 makes a secure call to Lync End Point that has call forwarded to PSTN End Point2	Pass	
408258	Device generates INVITE with Replaces and Referred-By headers when it receives a REFER request	Pass	
408254	Device includes REFER in ALLOW header in INVITE sent to Mediation Server	Pass	
408259	Device maintains the original session when rejecting a call transfer with REFER	Pass	
408257	Device supports Hairpin Elimination for Blind Transfer with REFER	Pass	
408260	Device supports Hairpin Elimination for secure Blind Transfer with REFER	Pass	
408255	PSTN End Point1 calls Lync End Point and Lync End Point Blinds Transfers the call to PSTN End Point2	Pass	
408256	PSTN End Point1 makes a secure call to Lync End Point and Lync End Point Blinds Transfers the call to PSTN End Point2	Pass	
408263	Device does not drop the call when Consultative Transfer by Lync End Point to second PSTN End Point fails	Pass	
408264	Device supports Hairpin Elimination for Consultative Transfer with REFER	Pass	
408265	Device supports Hairpin Elimination for secure Consultative Transfer with REFER	Pass	
408261	PSTN End Point1 calls Lync End Point and Lync End Point Consultative Transfers to PSTN End Point2	Pass	
408262	PSTN End Point1 makes a secure call to Lync End Point and Lync End Point Consultative Transfers to PSTN End Point2	Pass	
408213	Lync End Point1 calls Lync End Point2 and escalates the call to a conference, inviting PSTN End Point and later removing it	Pass	
408214	PSTN End Point establishes a call with the Conference Auto Attendant	Pass	
408215	Conference call involving two Lync End Points and PSTN End Point,	Pass	



	Lync End Point puts the call on hold		
408309	Device distributes new calls among DNS configured Mediation Serversn	Pass	
408311	Device honors TTL when distributing new calls among DNS configured Mediation Servers	Pass	
408286	Device responds to OPTIONS as keep alive to Mediation Server over TCP	Pass	
408288	Device responds to OPTIONS as keep alive to Mediation Server over TLS	Pass	
408289	Device resumes sending calls to Mediation Server when it starts receiving OPTIONS response from that Mediation Server	Pass	
408292	Device routes calls from newly added Mediation Server after DNS cache is updated	Pass	
408287	Device sends periodic OPTIONS message as keep alive to Mediation Server	Pass	
408285	Device uses load balancing to distribute inbound calls among Mediation Servers in a cluster	Pass	
408290	Lync End Point establishes a call with PSTN End Point when interface between Device and Mediation Server1 goes down	Pass	
408291	PSTN End Point establishes a call with Lync End Point when interface of Mediation Server1 goes down	Pass	
408294	Device does not offer new calls to a failed Mediation Server	Pass	
408293	Device fails over incoming call to a second Mediation Server when the first Mediation Server times out	Pass	
408306	Device utilizes failover and does not offer new calls to a failed Mediation Server	Pass	
408058	PSTN End Point calls Lync End Point with Caller ID set to 'Anonymous' on Device	Pass	



408321	Device disconnects call when Mediation Server sends 408 Request Timeout for call from PSTN End Point	Pass	
408327	Device disconnects call when Mediation Server sends 501 Not Implemented for call from PSTN End Point	Pass	
408328	Device disconnects call when Mediation Server sends 606 Not Acceptable for call from PSTN End Point	Pass	
408319	Device generates 482 Loop Detected response to a call from Lync End Point when a loop is detected	Pass	
408325	Device generates 486 Busy Here response from a busy PSTN End Point	Pass	
408324	Device handles call from Lync End Point to a user that does not exist in the domain	Pass	
408318	Device processes 482 Loop Detected response from Lync End Point	Pass	
408326	Device processes 486 Busy Here response from a busy Lync End Point	Pass	
408317	Device processes 488 Not Acceptable Here response for unsupported codec from Mediation Server	Pass	
408322	Device processes 603 Decline from Lync End Point for a secure call	Pass	
408323	Device processes 603 Decline response from Lync End Point	Pass	
408320	Device rejects call from Lync End Point to PSTN End Point when the associated PRI line is down	Pass	
408329	Device responds with 488 Not Acceptable Here when Mediation Server offers a codec unsupported on the device	Pass	
408315	Device sends 414 Request-URI Too Long when unable to handle very long Request URI	Pass	
408316	Device times out after 180 seconds of no response from Lync End Point following 100 Trying	Pass	



Troubleshooting Tools

If you find that there are issues with call setup, signaling, etc. or have problems with the test cases, there are a few tools available for Windows Server, Lync Server, and the Net-Net ESD 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 some minor issues you may encounter.

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.

Event Viewer

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 a few areas in which one would use the Event Viewer for troubleshooting:

- The Enterprise Voice client
- The Lync Front End server
- Lync Mediation server

Net-Net ESD

The Net-Net ESD 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 Net-Net ESD Console:

```
ACME1A# reset sipd
ACME1A# notify sipd siplog
ACME1A# notify sipd debug
enabled SIP Debugging
ACME1A# notify all rotate-logs
```

Examining the log files.

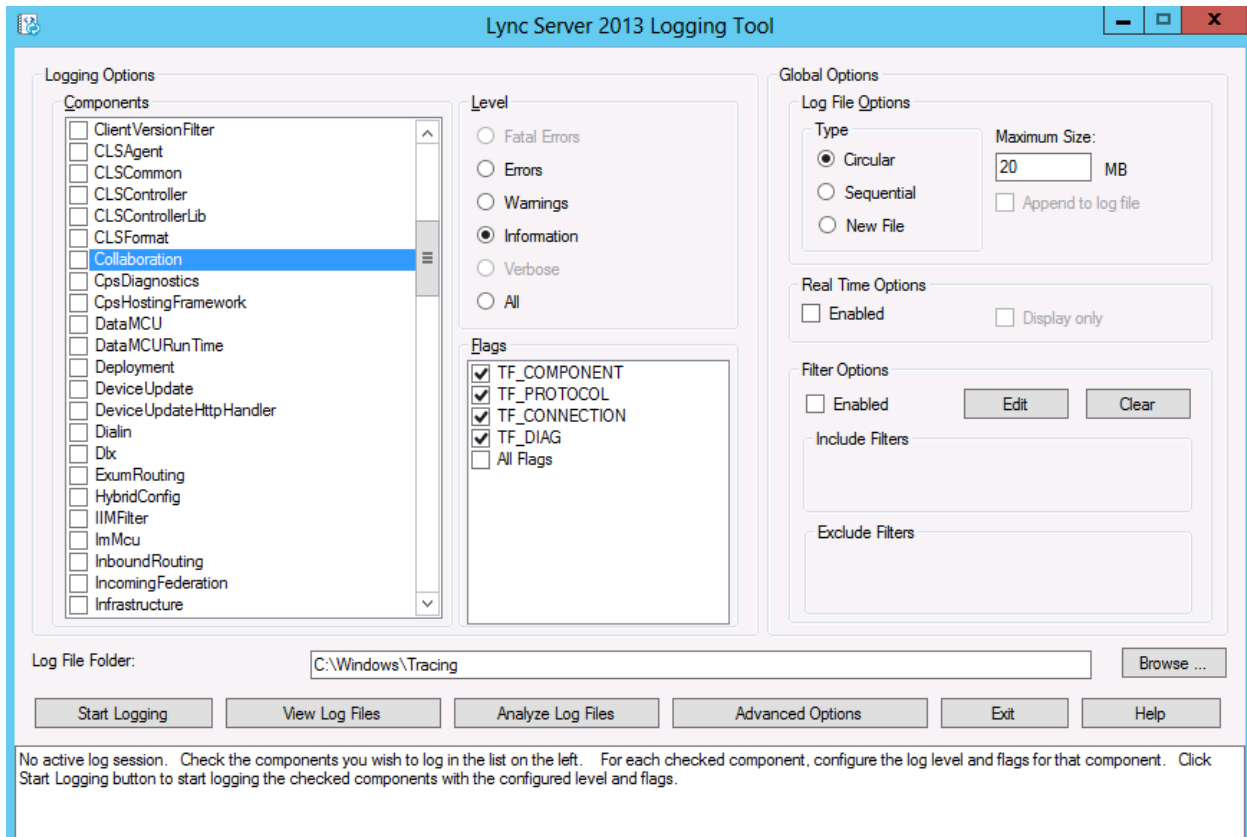
Note: You will FTP to the management interface of the Net-Net SBC with the username user and user mode password (the default is “acme”).

```
C:\Documents and Settings>ftp 192.168.5.24
Connected to 192.168.85.55.
220 ACME1A FTP 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.

Lync Server Logging Tool

The Lync Server 2013 Logging Tool provides internal traces and messaging between different Lync Server 2013 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 Lync Server 2013/Lync Server Logging Tool.





Appendix

Known Issues

No Ring Back Tone heard for inbound calls from PSTN to MS Lync through E-SBC

Recently, in some accounts where MS Lync and Acme Packet SBCs are deployed for enterprise voice and SIP trunk termination to an enterprise, there have been complaints of the PSTN caller hearing a silence when a call is placed from PSTN to a Lync user on the enterprise especially when Media Bypass is enabled on MS Lync

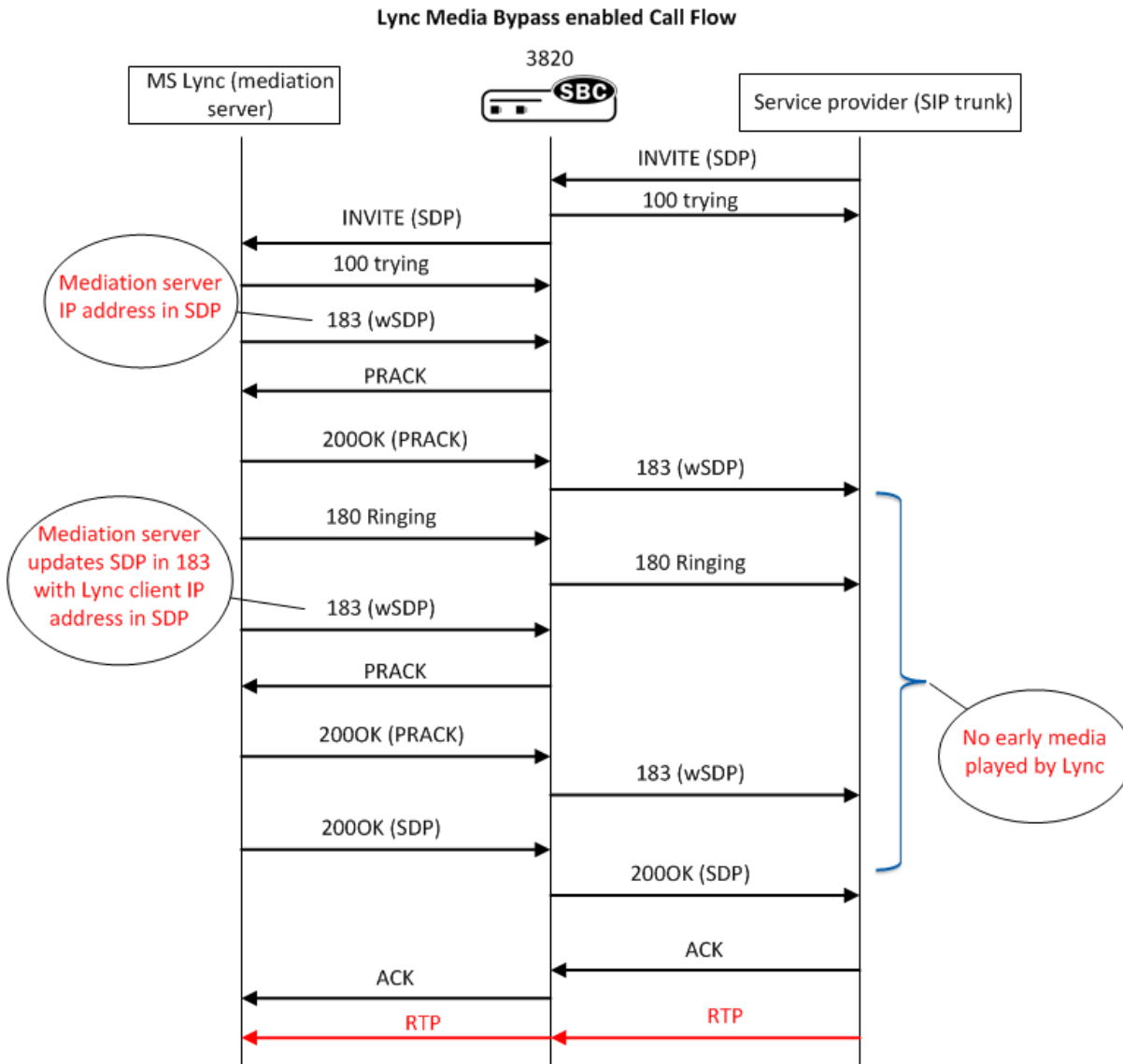
The configuration note below aims to explain this scenario briefly, steps taken to rectify this issue and proposed workaround by Acme Packet. The workaround is an interim solution while a permanent solution is being researched and developed by Acme Packet Engineering

Media Bypass

As explained earlier in the document, in order for Media Bypass to work, both Client and gateway (SBC) need to use the same RTP format, either SRTP (by default) or RTP. In default configuration of MS Lync, Lync client is required to use media encryption, so Media Bypass is mainly when media is encrypted (SRTP) and exchanged between Lync client and PSTN gateway (E-SBC).

Signaling between mediation server and SBC is a little different (Two 183s with SDP coming from mediation server) when media bypass is enabled on Lync.

The following is the call flow:



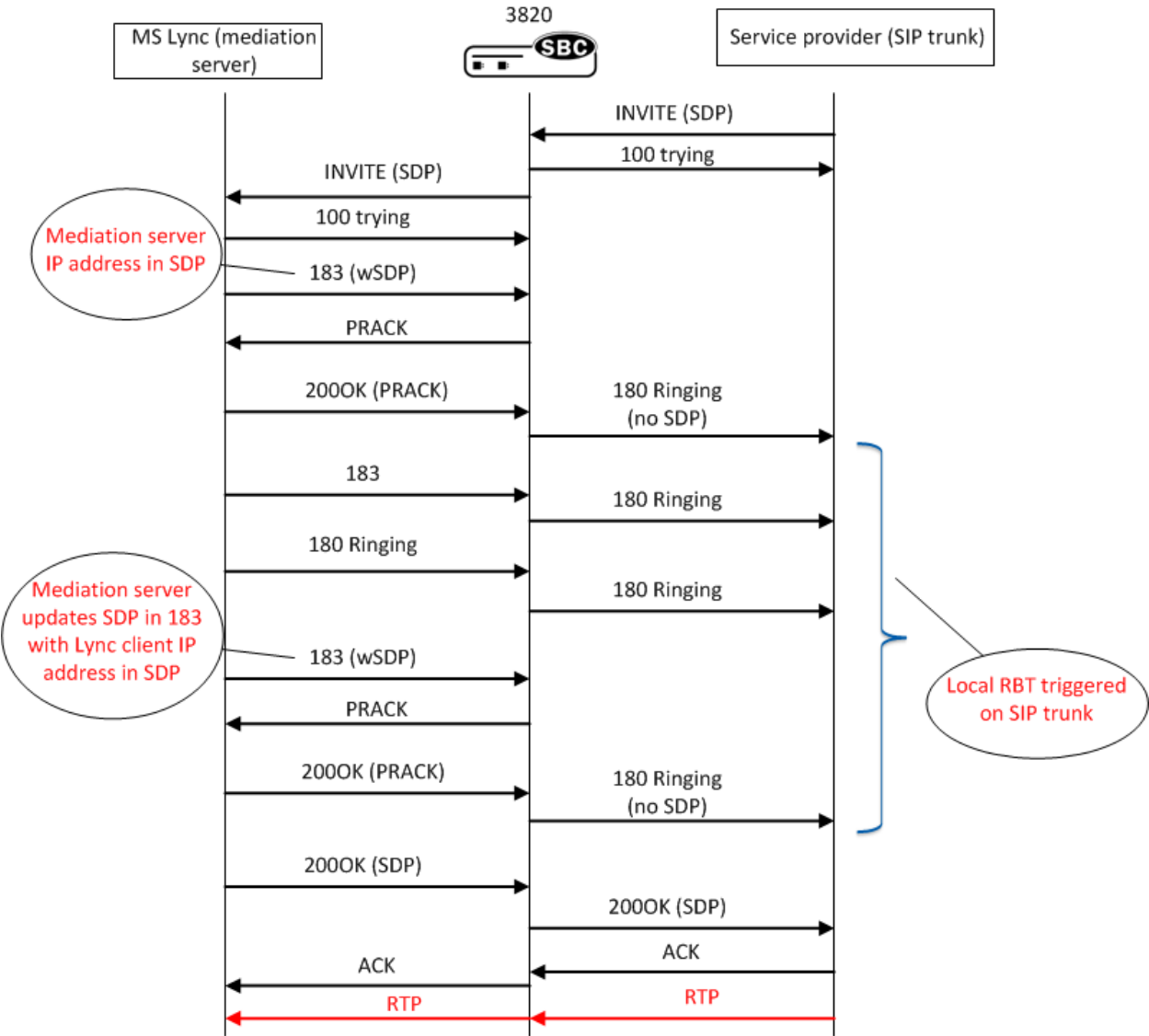
Note that after signaling 183 with SDP, Lync never plays any early media and expects gateway (E-SBD) to signal appropriately to the SIP Trunk provider to follow RFC 3960 and play local RBT. The second 183w SDP coming from Mediation server which is forwarded to the SIP trunk and stops the local RBT which was started after 180 Ringing was sent, hence PSTN caller would hear a silence before Lync client answers call.

Acme Packet Work Around

The interim solution is to present 180 ringing (convert all 183s on lync side to 180 ringing towards SIP trunk and strip the SDP) to trigger RBT in ISUP. The call flow is modified with the help of Acme Packet's robust Sip Manipulation and Sip Response Map features to the following:



Lync Media Bypass Call Flow Modified with Acme Packet Workaround



Configuration Changes required

In-manipulation Forearlymedia to be applied on Lync facing sip-interface

=====

```

sip-manipulation
    name Forearlymedia
    description
    split-headers
  
```




join-headers

header-rule

name	delsupported
header-name	Supported
action	delete
comparison-type	case-sensitive
msg-type	request
methods	INVITE
match-value	
new-value	

header-rule

name	addrequireinINVITE
header-name	Require
action	add
comparison-type	case-sensitive
msg-type	request
methods	INVITE
match-value	
new-value	100rel

header-rule

name	formod183
header-name	From
action	sip-manip
comparison-type	case-sensitive
msg-type	any
methods	
match-value	
new-value	Stripsdp183

sip-manipulation

name	Stripsdp183
description	For incoming 183 from Lync, strip SDP
split-headers	
join-headers	
header-rule	
name	check183
header-name	@status-line
action	store
comparison-type	pattern-rule
msg-type	any
methods	
match-value	
new-value	
element-rule	
name	is183
parameter-name	
type	status-code
action	store
match-val-type	any
comparison-type	pattern-rule
match-value	183
new-value	
header-rule	
name	delSDP
header-name	Content-Type
action	manipulate
comparison-type	case-insensitive



msg-type	any
methods	
match-value	\$check183.\$is183
new-value	
element-rule	
name	del183SDP
parameter-name	application/sdp
type	mime
action	delete-element
match-val-type	any
comparison-type	boolean
match-value	
new-value	
header-rule	
name	delContentType
header-name	Content-Type
action	manipulate
comparison-type	boolean
msg-type	any
methods	
match-value	\$check183.\$is183
new-value	
element-rule	
name	delCT
parameter-name	*
type	header-param
action	delete-header
match-val-type	any



comparison-type	case-sensitive
match-value	
new-value	

SIP Response Map to be applied on SIP Trunk facing Sip-interface

=====

response-map	
last-modified-by	admin@10.0.221.18
last-modified-date	2012-06-04 11:14:17
name	change183to180
entries	183 -> 180 (Ringing)
sip-interface	
state	enabled
realm-id	pstn
description	SIP Trunk-facing interface
sip-port	
address	10.10.1.4
port	5060
transport-protocol	TCP
tls-profile	
allow-anonymous	agents-only
...	
response-map	change183to180