APPLICATION NOTE: AN10440

Integrating ShoreTel with Microsoft Lync via AudioCodes IP to IP (SIP)
Microsoft Lync 2010 and AudioCodes Mediant 1000 Media Gateway

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Author Ed Hayes and Kurt Kruger
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Overview

The purpose of this application note is to illustrate integrating the ShoreTel UC system with Microsoft (MS) Lync Server 2010 for delivering voice calls between the two systems. This integration allows Lync audio conferencing calls to and from external callers via trunks on the ShoreTel system. The same steps are required to enable users on either system to place voice calls to one another (extension to extension dialing).

This application note does not provide details on the following integration features:
- MS Outlook Voicemail integration for client-side unified messaging
- MS Outlook Calendar integration for automated call handling mode changes
- MS Outlook Calendar integration for automated conference bridge details within appointments
- MS Outlook Contact importing for easy dial-by-name and dial-by-company
- TAPI integration for dialing directly from within Outlook contacts
- MS Exchange integration for server-side unified messaging
- MS Lync integration for IM, presence, or remote call control

This document focuses exclusively on the integration of MS Lync “Enterprise Voice” capabilities with the ShoreTel UC system via the AudioCodes Mediant media gateway products. Refer to other application notes and product documentation for details on all of the other integration methods. Please see the ‘References and Resources” section at the end of this document for a complete listing of related documentation and other configuration resources.

MS Lync enterprise voice access to the public switched telephony network (PSTN) can be delivered via ShoreTel voice switches and the AudioCodes Mediant family of media gateways. Microsoft Lync audio conferencing users and ShoreTel IP phone users can place, transfer, and conference calls between the two systems.

Required Components

1) AudioCodes Mediant 1000, 2000, or 3000 media gateway
   - One Media Processing Module (MPM) card for every six concurrent calls through the Mediant 1000 gateway
   - Two IP to IP “Feature Key” licenses for each concurrent call through the Mediant 1000, 2000 or 3000 gateway

2) Microsoft Lync Server 2010 with Mediation Server server role configured

3) One or more ShoreTel Voice Switches with available SIP trunk capacity
Supported and Tested Versions

This document is written based on testing with the following versions of software:

- ShoreTel 12 build 17.10.1730.0
- MS Lync Server 2010
- AudioCodes Mediant 1000 version 6.20A.048.002

Functionality differences based on future software updates from ShoreTel, AudioCodes, and Microsoft will be reflected as needed via updates to this document and other supporting resources. See the ‘References and Resources” section at the end of this document.

Important Disclaimer

This document is for informational purposes only and is provided “AS IS”. Microsoft and its partners cannot verify the accuracy of this information and take no responsibility for the content of this document. To the extent permitted by law, Microsoft makes no warranties of any kind, disclaims all express, implied and statutory warranties, and assumes no liability to you for any damages of any type in connection with the content of this document.

Intended Audience

The information provided in this document has been provided by Microsoft Partners or equipment manufactures and is provided “AS IS”. This document contains information about how to modify the configuration of your PBX or VoIP gateway. Improper configuration may result in the loss of service of the PBX or gateway. Microsoft is unable to provide support or assistance with the configuration or troubleshooting of components described within. Microsoft recommends readers to engage the service of a Microsoft Lync specialist or the manufacturers of the equipment described within to assist with the planning and deployment of Lync 2010.

AudioCodes Mediant Media Gateways

AudioCodes offers the enterprise customer a key component in the unified telephony network: the Media Gateway that connects the ShoreTel system with MS Lync 2010. AudioCodes Mediant media gateways are cost-effective VoIP gateways packaged in a stackable 1U or 2U chassis; they are designed to interface between the traditional PSTN and IP networks, or between two IP-based systems such as ShoreTel and MS Lync.
Initial Configuration

The configuration information below shows examples for configuring ShoreTel and the AudioCodes gateway. Even though configuration requirements can vary from setup to setup, the information provided in these steps, along with that found in the ShoreTel Planning and Installation Guide and the documentation provided by AudioCodes and Microsoft, should prove to be sufficient. However every design can vary and some may require more planning than others.

ShoreTel Configuration

This section describes the ShoreTel system configuration to support SIP Trunking. The section is divided into general system settings and trunk configurations (both group and individual) needed to support SIP Trunking.

ShoreTel System Settings – General
The first settings to address within the ShoreTel system are the general system settings. These configurations include the Call Control, the Site and the Switch settings. If these items have already been configured on your system, skip this section and go on to the “ShoreTel System Settings – Trunk Groups” section below.

Call Control Settings
The first settings to configure within ShoreTel Director are the Call Control Options. To configure these settings for the ShoreTel system, log into ShoreTel Director and select “Administration” then “Call Control” followed by “Options” – the “Call Control Options” screen will then appear (Figure 2).
Within the “Call Control Options” SIP parameters; confirm that the appropriate settings are made for the “Realm”, “Enable SIP Session Timer” and “Always Use Port 5004 for RTP” parameters.

The “Realm” parameter is used in authenticating all SIP devices. It is typically a description of the computer or system being accessed. Changing this value will require a reboot of all ShoreTel switches serving SIP extensions. It is not necessary to modify this parameter to get the ShoreTel IP PBX system functional with AudioCodes gateway.

**Step 1** Verify that the “Enable SIP Session Timer” box is checked (enabled).

**Step 2** Set the Session Interval Time to the recommended setting of 3600 seconds.

**Step 3** Select the appropriate refresher (from the pull down menu) for the SIP Session Timer. The “Refresher” field will be set either to “Caller (UAC)” [User Agent Client] or to “Callee (UAS)” [User Agent Server]. If the “Refresher” field is set to “Caller (UAC)”, the Caller’s device will be in control of the session timer refresh. If “Refresher” is set to “Callee (UAS)”, the device of the person called will control the session timer refresh.

**Step 4** Verify the “Voice Encoding and Quality of Service”, specifically the “Media Encryption” parameter. Make sure this parameter is set to “None”, otherwise you may experience one-way audio issues. Please refer to *ShoreTel Administration Guide* for
additional details on media encryption and the other parameters in the “Voice Encoding and Quality of Service” area.

**Step 5** Disable (uncheck) the “Always Use Port 5004 for RTP” parameter if checked; it is required for implementing SIP trunks between ShoreTel systems only. For SIP configurations, Dynamic User Datagram Protocol (UDP) must be used for RTP Traffic. If the parameter is disabled, Media Gateway Control Protocol (MGCP) will no longer use UDP port 5004; MGCP and SIP traffic will use dynamic UDP ports (Figure 3).

![Figure 3 – Call Control Options Settings](image)

Once this parameter is unchecked, make sure that “everything” (IP Phones, ShoreTel Voice Switches, ShoreTel Server, Distributed Voice Mail Servers / Remote Servers, Conference Bridges and Contact Centers) is “fully” rebooted – this is a “one time only” item. By not performing a full system reboot after changing this setting, one-way audio may occur during initial testing.

**Step 6** Be sure to save your changes before leaving this screen by clicking Save at the top of the page.

**Sites Settings**
The next settings to address are the administration of sites. These settings are modified under the ShoreTel Director by selecting “Administration” then “Sites”.

This selection brings up the “Sites” screen.
Step 1  Within the “Sites” screen select the name of the site to configure. The “Edit Site” screen will then appear.

The only changes required to the “Edit Site” screen are to the “Admission Control Bandwidth” and “Intra-Site / Inter-Site Calls” parameters (Figure 4).

![Figure 4 – Site Bandwidth settings](image)

Step 2  Set the appropriate Admission Control Bandwidth for your network. Please refer to the *ShoreTel Planning and Installation Guide* for additional information on setting Admission Control Bandwidth for your network. Admission Control Bandwidth defines the bandwidth available to and from the site. This is important as SIP trunk calls will be counted against the site bandwidth.

*Note:* Bandwidth of 2046 kbps is just an example.

Step 3  Configure the "Inter-Site Calls” option for "Very Low Bandwidth Codecs." By default "Very Low Bandwidth Codecs" contains two codecs, G.729 and G.711u, with G.729 being the primary codec of choice. The “Inter-Site Calls” parameter defines which codecs will be used when establishing a call with AudioCodes – the preferred codec choice is G.729.
Note: Please do not modify the "Very Low Bandwidth Codecs" codec list.

**Step 4**  Save changes before leaving this screen by clicking **Save** at the top of the page.

**Switch Settings - Allocating Ports for SIP Trunks**

The final general settings to configure are the ShoreTel switch settings. These changes are modified by selecting “Administration” then “Switches” followed by “Primary” in ShoreTel Director (Figure 5).

![Figure 5 – Administration Switches](image)

This action brings up the “Switches” screen. From the “Switches” screen choose the name of the switch to configure for SIP trunks. The “Edit ShoreTel Switch” screen will appear.

**Step 1**  Within the “Edit ShoreTel Switch” screen, select the desired number of SIP Trunks from the ports available (Figure 6).
Each port designated as a Port Type of “SIP Trunk” enables the support for five individual SIP trunks. The AudioCodes Mediant 1000 can be configured for up to 120 SIP trunks. Each trunk can support one concurrent call between the ShoreTel system and the MS Lync 2010 system. Determine the desired capacity of the interconnection between the two systems and configure the necessary resources as required, then proceed to the next section.

Step 2  Be sure to save your changes before leaving this screen by clicking Save at the top of the page.

ShoreTel System Settings – Trunk Groups
ShoreTel Trunk Groups only support Static IP Addresses for Individual Trunks. In trunk planning, the following needs to be considered. AudioCodes gateway interfaces should always be configured to use a “static” IP Address.

The settings for Trunk Groups are changed by selecting “Administration”, then “Trunks” followed by “Trunk Groups” within ShoreTel Director (Figure 7).
Step 1  From the pull down menus on the “Trunk Groups” screen, select the site desired and select the “SIP” trunk type to configure.

Step 2  Click on the “Go” link from “Add new trunk group at site”. The “Edit SIP Trunk Group” screen will appear.
Step 3 Enter your preferred name for the new trunk group. In the example in Figure 8, the name “AudioCodes SIP to SIP” has been created.

Step 4 Enable (check) the setting of the “Teleworker” parameter. Enabling the “Teleworker” parameter will configure the trunk group for inter-site calls, which was configured earlier for “Very Low Bandwidth Codecs” which has G.729 as the primary codec choice. Once this parameter is enabled, it will count against the site bandwidth.

Step 5 The “Enable SIP Info for G.711 DTMF Signaling” parameter should not be enabled (checked). Enabling SIP info is currently only used with SIP tie trunks between ShoreTel systems.

Step 6 The “Profile:” parameter is should be left at a default setting of “_SystemTrunk”; It is not necessary to modify this parameter when connecting to the AudioCodes gateway.

Step 7 The “Enable Digest Authentication” parameter defaults to “<None>” and modification is not required when connecting to the AudioCodes gateway.

The next item to change in the “Edit SIP Trunks Group” screen is to make the appropriate settings for the “Inbound:” parameters (Figure 9).
Step 8  Within the “Inbound:” settings, ensure the “Number of Digits from CO” is set to match what the ShoreTel SIP trunk switch will be receiving from AudioCodes gateway and ensure the “DNIS” or “DID” box is enabled (checked), along with the Extension parameter.

Step 9  We recommend that the Tandem Trunking parameter be enabled (checked) otherwise transfers to external telephone numbers will fail via SIP trunks. For additional information on this parameter please refer to the *ShoreTel Planning and Installation Guide*.

The next item to change in the “Edit SIP Trunks Group” screen is to make the appropriate settings for the “Onbound:” parameters (Figure 10).
Step 10

Enable (check) the “Outbound” parameter and define a Trunk “Access Code” and “Local Area Code” as appropriate.

In the “Trunk Services:” area, make sure the appropriate services are enabled or disabled based on your needs. In general, we are only using this trunk group to dial the off system extensions to reach the Lync audio conferencing bridge or softphone users.

The last parameter determines if the call is sent out as <unknown> or with caller information (Caller ID). User DID will impact how information is passed out to the SIP Trunk group.

The final parameters for configuration in the Trunk Group are “Trunk Digit Manipulation” (Figure 11).
Step 11 Enable (check) the “Dial in E.164 Format” parameter to match the format MS Lync expects. and define a Trunk “Access Code” and “Local Area Code” as appropriate.

Next you must create the Off System Extension (OSE) range that will be used to represent the Lync audio conferencing bridge or Lync softphone users. An OSE is required for every Lync enterprise voice endpoint that will be using the ShoreTel system.

Step 12 Click the Edit button next to Off System Extensions:. The Off Systems Extension Range dialog is displayed (Figure 12)
Figure 12 – Off System Extension Ranges

**Step 13**  Click New and define the first range for the extensions that will represent the Lync enterprise voice endpoints on the ShoreTel system.

**Step 14**  Click OK to save the first range and repeat if necessary to create sufficient extensions for all your Lync endpoints.

**Step 15**  After all your setting changes are made to the “Edit SIP Trunk Group” screen, click the “Save” button at the top of the page.

**ShoreTel System Settings – Individual Trunks:**
This section covers the configuration of the individual trunks. Select “Administration”, then “Trunks” followed by “Individual Trunks” to configure the individual trunks.

The “Trunks by Group” screen is used to change the individual trunks settings that appear (Figure 13).

Figure 13 – Trunks by Group
Step 1  Select the site for the new individual trunk(s) to be added and select the appropriate trunk group from the pull down menu in the “Add new trunk at site” area. In this example, the site is “Sunnyvale-HQ” and the trunk group is “AudioCodes SIP to SIP”, as created above.

Step 2  Click on the “Go” button to bring up the “Edit Trunk” screen.

Step 3  From the individual trunks “Edit Trunk” screen, input a name for the individual trunks, select the appropriate switch, select the SIP Trunk type and input the number of trunks. When selecting a name, the recommendation is to name the individual trunks the same as the name of the trunk group so that the trunk type can easily be tracked.

Step 4  Select the switch upon which the individual trunk will be created. For the “Sip Trunk Type”, the only option available is “Use IP Address”, enter the IP address of the AudioCodes gateway.

Step 5  Select the number of individual trunks desired (each one supports “one” audio path – example if 10 is configured, then 10 audio paths can be active at one time).

Note: Individual SIP Trunks cannot span networks. SIP trunks can only terminate on the switch selected. There is no failover to another switch. If you want to build a redundant configuration, two trunk groups will be needed with each pointing to another AudioCodes gateway – just the same as if PRI were being used.

Step 6  Once all changes are complete, select the “Save” button to commit changes.

Depending on your environment you may have to create a DNIS map on the primary PSTN trunk(s) coming into the ShoreTel system for the enterprise voice user since they are not defined in ShoreTel Director. See example below (Figure 14).

![Figure 14 – DNIS Digit Map](image-url)
After setting up the trunk groups and individual trunks, refer to the *ShoreTel Planning and Installation Guide* to make the appropriate changes for the User Group settings. This completes the settings for the ShoreTel system side.

**ShoreTel Technical Support**
In the event that you have problems with the ShoreTel system you may contact ShoreTel Technical Assistance Center at +1 (800) 742-2348 (Toll Free) or +1 (408) 331-3313 (International). A support contract must be in place before any assistance will be provided.

**AudioCodes Configuration**

Please refer to the AudioCodes Mediant 1000 MSBG Installation Manual Ver 6.20A.048.002 for detailed instructions on *Setting up the Device* and *Setting the IP Address* located in Chapters 2 & 3.

**Set up the AudioCodes Gateway**

The Mediant 1000 includes an embedded Web server (Web interface), providing a user-friendly graphical user interface (GUI) for configuring PSTN gateway-related functionality. The IP address used for accessing this Web interface must be changed to suit the networking scheme in which your Mediant 1000 is deployed. Before you can configure the Mediant 1000, you need to first access it with the default VoIP / Management LAN IP address, as described below.

**Step 1**
Connect one of the LAN ports on the CRMX module of the Mediant 1000 directly to a PC, using a straight-through Ethernet cable.

**Step 2**
Ensure that your PC is configured to automatically obtain IP addresses. The Enhanced gateway embedded DHCP server (enabled by default) allocates an IP address to the PC when connected to it.

**Step 3**
Open a standard Web browser and in the URL address field, enter the Enhanced gateway default VoIP / Management LAN IP address i.e., '192.168.0.2':

**Step 4**
Log in to the AudioCodes Mediant 1000 with the appropriate credentials. The screen will display the Mediant Gateway home page (Figure 15).
Open the ‘IP Settings’ page under Configuration tab > Network Settings menu > IP Settings page item (Figure 16).

**Step 5** Define the device’s IP address, subnet mask, and default Gateway IP address so that they correspond to your network IP scheme.

**Step 6** Click Submit. You will now lose connectivity to the gateway.

**Step 7** Reset your PC’s IP address so that it can connect to the Mediant 1000 using the new IP address, then connect to it via your browser.

**Step 8** Save your settings to the flash memory by clicking Burn.

**Step 9** Reset the device by clicking Device Actions > Reset and then Reset (Figure 16).

*Figure 16 – AudioCodes Reset*

*Note:* This takes approximately three minutes to complete.
Step 12  Disconnect your PC from the Mediant gateway.

Step 13  Restore your PC’s IP address and subnet mask to what they originally were. If necessary, restart your PC.

Step 14  Reconnect the Mediant gateway and your PC (if necessary) to the network.

You should now be able to access the Mediant via the embedded web server with its new assigned IP address.

Continue to the next section to customize the AudioCodes configuration file for your environment.

**Configure the AudioCodes INI file**

Step 1  Click **Management Configuration > Management Settings** to configure a Syslog server (**Figure 17**) to aid in debugging the communication between the Mediant and other devices on the network. The Syslog server can be the supplied AudioCodes proprietary Syslog server (ACSyslog) or any third-party Syslog servers such as Kiwi Enterprises (www.kiwisylog.com/).

![Figure 17 – Enabling Syslog Server](image)
Step 2  Click **Software Update > Software Upgrade Key** to load your AudioCodes software license key (Figure 18). Note that to enable SIP connections between the ShoreTel system and the Mediant, your license key must include IP2IP=Y in the Control Protocols list.

![Software Upgrade Key Status](image)

**Figure 18 – Loading a software license key**

Step 3  Click **IP Media > IP Media Settings** to verify the number of media channels you have configured (Figure 19). In this example we’ve configured 16 channels. Note that a maximum of 96 channels with Microsoft SRTP can be enabled on the Mediant 1000 for concurrent call through the gateway.

![IP Media Settings](image)
Step 4  Click Protocol Configuration > Proxies, Registration, IP Groups > IP Group Table to establish the SIP trunk between the Mediant and the ShoreTel system. Enter the same IP address here in the Description field for the ShoreTel Voice Switch you selected on the Edit Trunks panel when you added the SIP trunk to the ShoreTel system. Note the Index number for this IP Group is 1 in this example (Figure 20).

![Figure 20 – IP Group for SIP trunk](image)

Step 5  Add a second IP Group (Index number 2 in this example) for the Lync Mediation Server or pool. Enter the IP address or FQDN here in the Description field for the Mediation Server. (Figure 21)
Step 6  Click Configuration Tab > Control Network > Proxy Sets Table to enter the Proxy Address for the SIP trunk to on the ShoreTel Voice Switch. Use the same IP address as above for the IP Group table and note that the UDP transport flag must be set as well. (Figure 22)
Step 7 Add a second entry in the Proxy Address table for the Mediation Server using its IP address or FQDN. (Figure 23). Set the TCP transport type flag as well.
Step 8  
Click VoIP> GW and IP to IP > Manipulations> Dest Number IP->Tel table (Figure 24). Click Add and enter the following in order to remove the leading + on ext to ext routing.

- Destination Prefix   +
- Source Prefix         *
- Source IP address     *
- Stripped Digits from left  1
- Stripped Digits from right  0
- Prefix to Add         blank
- Suffix to Add          blank
- Number of Digits to Leave  255

Leave NPI and TON as default “Not Configured”.

Step 9  
Click VoIP> GW and IP to IP > Manipulations>  Source Number IP->Tel table (Figure 25). Click Add and enter the following in order to remove the leading + on ext to ext routing.

- Destination Prefix   +
- Source Prefix         *
- Source IP address     *
Click **Burn**

Your Mediant 1000 is now ready to interconnect ShoreTel phone users and Lync softphone users for extension to extension dialing or to allow Lync audio conferencing to access the PSTN through the ShoreTel system. Proceed to the next section to set up the connection on the Microsoft Lync Mediation Server side.

### Microsoft Lync Configuration

**Note:** Microsoft recommends readers engage the services of a Microsoft Lync specialist to assist with the planning and deployment of Lync Server 2010 and Enterprise Voice configuration.

This topic assumes that you have set up at least one Enterprise Edition Front End pool or Standard Edition server in at least one central site as described in Microsoft’s Lync deployment documentation. This section also assumes that you have already fully deployed
Enterprise Voice, including Dial Plans, Voice Policies, and PSTN usage as required. Deploying Lync Enterprise Voice is beyond the scope of this document.

**Add AudioCodes Gateway to the Topology**

**Step 1** Connect to the MS Lync 2010 Server and run Topology Builder. Open existing topology you'll have to give the file a name and save it. Your existing running deployment should open. *(Figure 26)*

![Figure 26 – Lync Topology Builder](image)

**Step 2** Once open select the site at which you plan to deploy. If Lync 2010 is currently deployed, the front end pool will already be there. For the purpose of this technical brief we're only concerned with the Mediation Server tab.

**Step 3** Click Edit Properties *(Figure 27)* and you'll see there are four configurable tabbed areas that allow for user defined configuration. Depending on how Lync is deployed in your environment, you may have a collocated mediation server or a standalone mediation server.
Step 4 If using a collocated mediation server, make sure collocated box is enabled and checked. Listening ports are TLS:5067 and TCP:5068. You’ll see two configurable gateway boxes. The top box shows gateways that aren’t associated with any Mediation Server. Click Add to associate them with this Mediation Server. The bottom one shows which gateways are already associated with this Mediation Server.

Step 5 For the purpose of this technical brief, choose the second, click New and add the IP address of the Mediant 1000 gateway. Under SIP Transport protocol, click TCP and specify the listening port as 5068 and click OK.

Step 6 Navigate to PSTN gateways (Figure 28) then from the Actions window select New IP/PSTN gateway.

Step 7 Define the gateway IP address and TCP listening port 5068 then click OK.
Step 8  

Next you’ll publish the Lync topology by selecting Publish from the Actions > Topology menu (Figure 29).
Once the topology is successfully published, close the Topology Builder and move on to the next section.

Configure Lync Voice Routes and Normalization Rules

Lync Global Dial Plan is the default dial plan installed with the product. You can edit the global dial plan, but you cannot delete it. This dial plan applies to all Enterprise Voice users, groups, and contact objects in your deployment unless you configure and assign a dial plan with more specific scope.

A dial plan is a named set of normalization rules that translates phone numbers for a named location, individual user, or contact object into a single standard (E.164) format for purposes of phone authorization and call routing.

Lync Normalization Rules define how phone numbers expressed in various formats are to be routed for each specified location, user, or contact object. The same dial string may be
interpreted and translated differently depending on the location from which it is dialed and the person or Contact objects making the call.


In order to route Enterprise Voice calls between ShoreTel and Lync system via the AudioCodes gateway, you will need to configure a Lync Voice Route which will associate your Dial Plan with the AudioCodes Gateway IP address.

Before you define outbound call routes, you should complete the following steps:

- Deploy one or more PSTN gateways or SIP trunking connections and Lync Server 2010 Mediation Servers.
- Create dial plans as needed for sites, individuals, and Contact objects.

In addition, to enable outbound call routing, you must create and assign one or more voice policies. This task can be done either before or after you define outbound call routes.

For each route, you must specify:

- A name by which it can be readily identified.
- An optional description in cases where the name alone may not be sufficient to describe the route.
- The regular expression matching pattern that identifies the destination phone numbers to which the route is applied, along with exceptions to which the matching pattern is not to be applied.
- The FQDN of one or more the gateways that you want to assign to the route.
- The PSTN usage records that users must have in order to call numbers matching the destination phone number regular expression.

More details on setting up Voice Routes can be found at http://technet.microsoft.com/en-us/library/gg412757

**Step 1**  On the Lync server, launch the Lync Server Control Panel. Select Voice Routing on the left and then select Route at the top of the menu (Figure 30).

**Step 2**  Select New and give the new route a name and note that the description is optional. At the bottom click the Add button to add your gateway you created earlier in the Topology Builder.
Step 3  Select OK then select Commit All.

This next section describes creating a **Lync Normalization Rule** which is required to prepend your area code and prefix to the ShoreTel Off System Extension (OSE) via the Lync Dial Plan so when a call from ShoreTel reaches Lync, it will be converted into required E.164 format. The example below is if you are using 4 digit extensions which also correspond to your OSE range in your ShoreTel SIP trunk group.
**Step 1**  Open a browser window, and then enter the Admin URL to open the Lync Server Control Panel. Select Voice Routing then select Dial Plan.

**Step 2**  Select Edit to open your Dial plan.

**Step 3**  In **New Normalization Rule**, type a name that describes the number pattern being normalized in **Name** (Figure 31). In this example, the name of the normalization rule is **4Digit Dialing to Lync**.

![Figure 31 – Lync Build Normalization Rule](image)

**Step 4**  Type a **Description** for the number pattern being normalized in **Name**. In this example, the description is **Add Prefix and Area Code from ST to Lync**.

**Step 5**  In **Build a Normalization Rule** box, enter values in the following fields:
Starting digits  Leave blank

Length  Specify the number of digits that equals the length of your extensions.

Digits to remove  (Optional) Specify the number of starting digits to be removed from dialed numbers you want the pattern to match.

Digits to add  Specify digits to be added to dialed numbers you want the pattern to match. This should be +1 with your area code and prefix. Example is +1408962

(Optional) If the normalization rule results in a phone number that is internal to your organization, select Internal extension.

(Optional) Enter a number to test the normalization rule and then click Go. The test results are displayed under Enter a number to test.

Step 6  Click OK to save the normalization rule then click OK to save the dial plan.

Step 8  On the Dial Plan page, click Commit, and then click Commit all.


Configure the Lync Clients

Step 1  Open the Lync 2010 Server Control Panel again.

Figure 32 – Launch Lync Server Control Panel
Note: The users to be configured must already be in Active Directory.

Step 2 Once logged into the control panel, navigate to Users then click User Search if the user is an existing Lync user that you want to configure.

Step 3 Open the Properties and navigate to the Telephony tab.

Step 4 Click the pull-down menu and choose Enterprise Voice (Figure 34) to configure the Lync user’s telephony settings, such as their phone number which Microsoft calls the “Line URI”. These Line URI entries must be in canonical format, such as tel:+14083316594.
Step 4  Click Commit and repeat this process for each user as required.

In this example, the Line URI `tel:+14083316594` maps to the ShoreTel off system extension (OSE) number, in this case x6594. Once you have all the users correctly configured, extension to extension calls on the ShoreTel system to their corresponding OSE will ring on the user’s Lync telephony endpoint, typically the Lync softphone.