

Installation and configuration of an Optimum SIP trunk with 3CX

Important considerations

The Optimum SIP trunk can be configured as a VoIP provider or as a true SIP trunk. The SIP Trunk mode is the recommended configuration due as it will be simpler to setup and there are no known security concerns because the interface resides on the same private LAN as the 3CX server.

Please note that the SIP termination device, the Edgewater appliance, has 4 LAN ports. Port 2 and 3 are disabled do not use them. Port 4 is to be used for the initial configuration. Do not connect port 4 to your LAN because the appliance will act as a DHCP server. You should connect a laptop and access the unit at 10.10.200.1 via a web browser (user: pbxinstall, password s1ptrunk) and setup a static LAN IP address and subnet mask as shown below. Please use a LAN IP that resides on the same subnet as the 3CX server. Once you click on submit, Port 1 will have this IP address. If you already have a DHCP server on your LAN, make sure that you do not click on "Enable DHCP server".

Optimum Business™

Configuration Menu

- Customer
 - LAN Settings
 - SIP Trunk Configuration
 - Diagnostics
 - System

LAN Settings [Help](#)

LAN Interface Settings:
 This address must be on the same subnet as the IP Address of the PBX

IP Address:

Subnet Mask:

DHCP Server:

Enable DHCP Server:

Start Address:

End Address:

The voice network must be connected to LAN port 1. LAN port 4 is dedicated to management and maintenance.

Once this step is completed, you can continue configuring the Edgewater appliance using Port 4 or you can connect Port 1 to the LAN and use any computer on your LAN to access it at the IP address that you specified at the previous step. The final step in configuring the Edgewater device is to setup the SIP service type.

In the dropdown window, please select 3CX as your PBX type, next if you intend to use the SIP Trunk configuration, select the "Passive connection" option and enter the PBX address in the window then click on submit.



Configuration Menu

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 - ▶ System

SIP Trunk Configuration

[Help](#)

Select your PBX:

- Passive connection using the local, private IP address of the PBX
This address must be on the same subnet as the IP Address that is specified for the LAN interface

PBX Address:

- Active connection using registration

User Id:

Password:

Convert Inband DTMF:

Status:

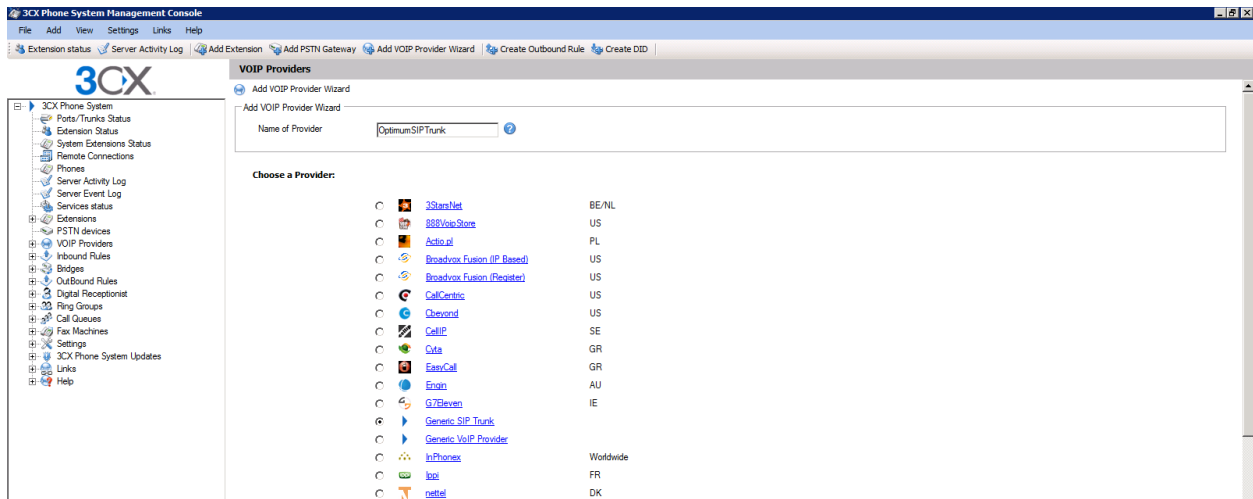
Trunk Status:

DID's

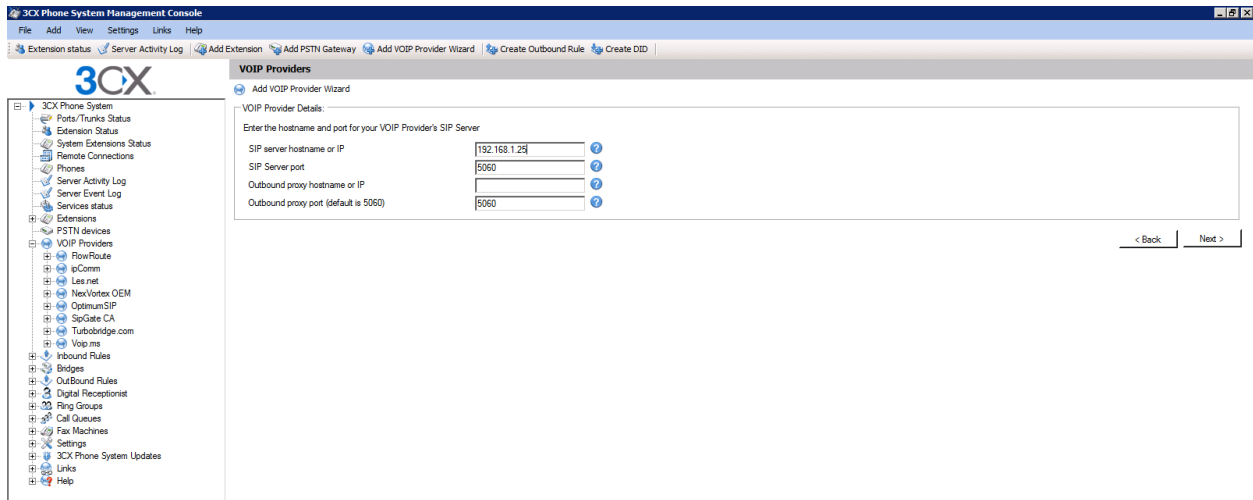
2035955320 (Pilot number)
2035955320
2035955321
2035955322

Setting up the 3CX server in SIP Trunk mode

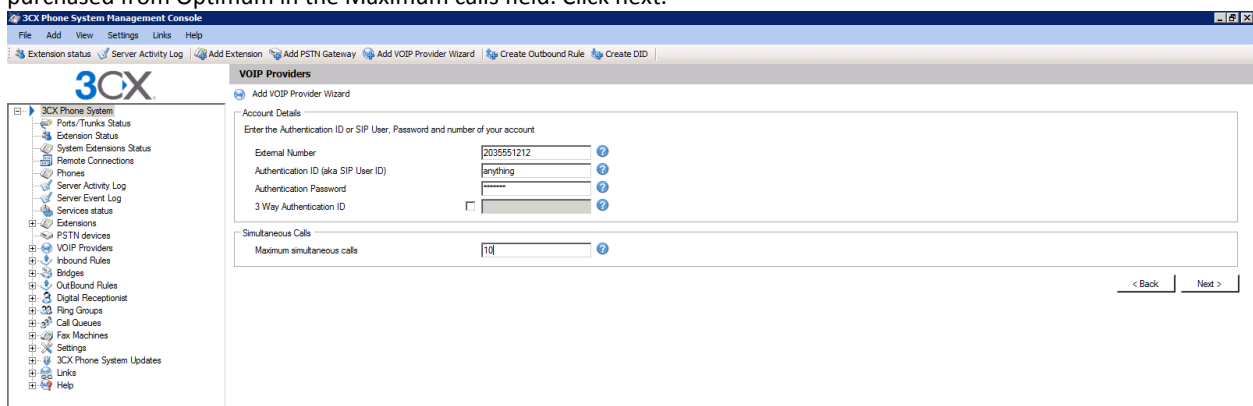
Click on VoIP providers, ADD a provider and the window below will appear. Pick any name that you'd like. Select "Generic SIP Trunk" and click on next.



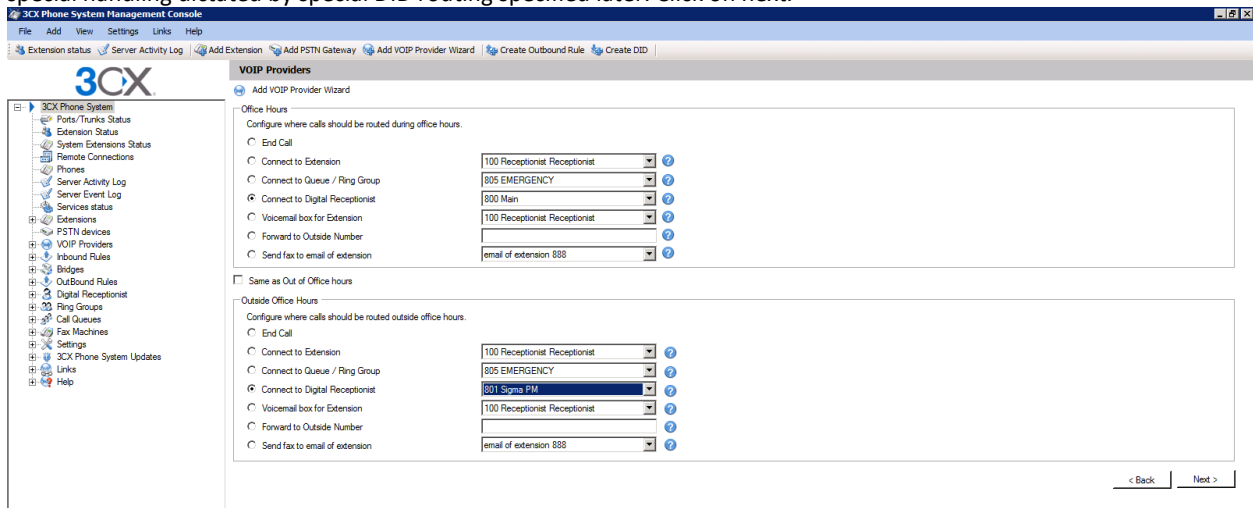
Enter the LAN IP address of the Edgewater appliance in the SIP server host name field. Make sure that the default ports remains 5060 and click on next.



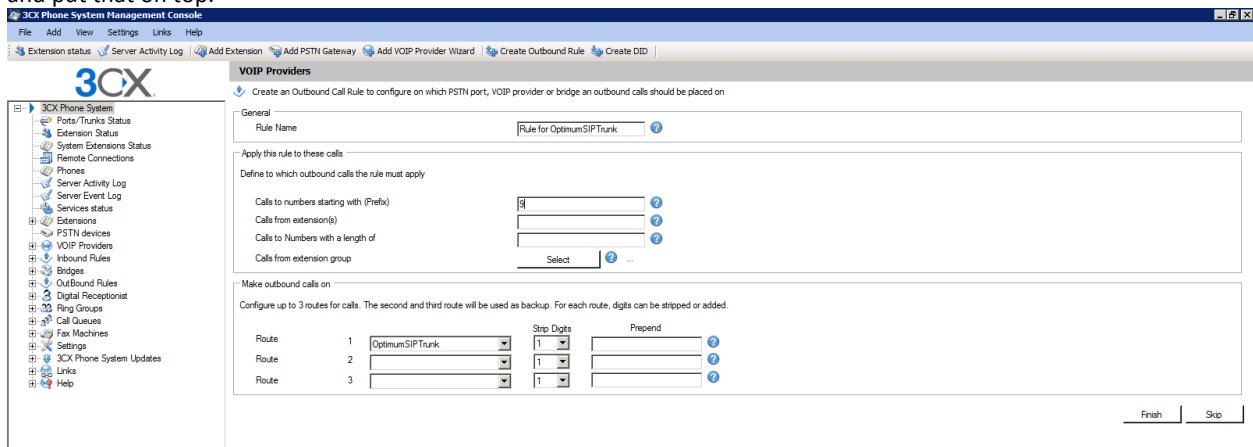
Enter your primary DID number or any unique number in the External number field. You can enter anything you please in the Authentication IP and password as it will not be used and will be ignored. Enter the number of sessions that you purchased from Optimum in the Maximum calls field. Click next.



Define the default inbound routing rules. Here you can select what happens with all incoming calls that do not match any special handling dictated by special DID routing specified later. Click on next.



Here you can specify a simple outbound rule to use in connection with your new SIP trunk. The simplest rule and the most popular is to specify that all calls starting with 9 will go out via the Optimum SIP trunk and the first digit is stripped. Please note that this is not sufficient in most setups and requires all outbound number to be dialed with a 9 followed by the full 10-digit number. It is also important to know that calls to 911 will not work because the server will strip the 9 and only send out 11 to the SIP trunk. Please make sure that if you use this type of configuration, you create a rule for 911 and put that on top.



Once you complete this step, please return in the configuration of the Optimum SIP trunk, select the advanced tab and uncheck the “PBX Delivers Audio” and check “Support Re-invites” and “Support Replace”. Then change the IP contact field to “Internal” as in the image below. This ensures proper DTMF operation.

It is extremely important that your phones be configured to deliver DTMF in audio, please modify the template as needed to remove RFC2833. A template for the Grandstream GXP-21XX and GXP-14XX is available as an add-on.

Edit VOIP Provider

Edit VOIP Provider settings and click OK or Apply to save changes

General | **Advanced** | Outbound Parameters | Inbound Parameters | Source ID | DID

Provider Capabilities
Configure options related to the SIP capabilities of your provider

Supports Re-Invite ?

Supports 'Replace' ?

PBX Delivers Audio ?

Switch on Secure RTP (SRTP) ?

Registration Settings
Configure options related to the SIP capabilities of your provider

Time between registration attempts (in seconds) ?

Require registration for: ?

Which IP to use in 'Contact' field for registration:

External(STUN resolved) ?

Internal ?

Specified IP ?

Codec priorities
Specify which codecs to use and according to which priority.

Available Codecs

GSM-FR
Speex
ILBC
G729
G722

Add >

< Remove

Assigned Codecs

G.711 U-law
G.711 A-law

Up

Down

Create DID based routing

Click on Create DID, you can use whatever you please in the DID/DDI name to help you recognize the incoming line on your phone display. In the DID/DDI number/mask please enter the 10-digit DID number you are adding. In the available ports, please select the SIP trunk that you previously configured for the Optimum SIP trunk and select the desired routing for office hours and outside office hours. Click OK.

3CX Phone System Management Console

File Add View Settings Links Help

Extension status Server Activity Log Add Extension Add PSTN Gateway Add VOIP Provider Wizard Create Outbound Rule **Create DID**

3CX

Add DID
Route calls to DID/DDI numbers directly to an extension

DID/DDI Name
Enter a DID or string to look for in the SIP 'to' field. Use wildcards (*) to match any digit for that entry. For example, entries 22444032 OR 2244403* will both match calls with a dialed number of +35722444032 in the 'to' field

DID/DDI Name ?

DID/DDI number/mask
Select from the drop-down below the type of inbound rule you want to create and enter a mask for this DID. You can use the * as a wildcard either before or after your mask.

Inbound Rule type ?

DID/DDI number/mask ?

Apply this rule to these ports
Select the Gateway you want this DID/DDI rule to be applied to. You can select on the whole gateway which will apply the rule to all the ports, or you can select individual ports.

Available ports

- ipComm
- Les.net
- NexVortex OEM
- FlowRoute
- Tubobridge.com
- Voip.ms
- SipGate CA
- OptimumSIP

Office Hours
Configure where calls to this DID/DDI should be routed during office hours.

End Call

Connect to Extension ?

Connect to Queue / Ring Group ?

Connect to Digital Receptionist ?

Voicemail box for Extension ?

Forward to Outside Number ?

Send fax to ?

Set up Specific Office Hours ?

Apply the same routing logic Outside of office hours

OK Cancel Apply

On the left side navigation bar, click on the SIP trunk for your Optimum SIP trunk, select the Source ID tab, check the "Source identification by DID" box and then click on "Add DID". Select all the DID that belong to this trunk and click on OK. You will need to do this every time you add a DID to this trunk.

3CX Phone System Management Console

File Add View Settings Links Help

Extension status Server Activity Log Add Extension Add PSTN Gateway Add VOIP Provider Wizard Create Outbound Rule Create DID

3CX

3CX Phone System

- Ports/Trunks Status
- Extension Status
- System Extensions Status
- Remote Connections
- Phones
- Server Activity Log
- Server Event Log
- Services status
- Extensions
 - PSTN devices
 - VOIP Providers
 - FlowRoute
 - pComm
 - Les.net
 - NexVox OEM
 - OptimusSIP
 - 2035955320
 - 2031234567
 - 2035955322
 - SipGate CA
 - TurboBridge.com
 - Vop.ms
 - Inbound Rules
 - Bridges
 - Outbound Rules
 - Digital Receptionist
 - Ring Groups
 - Call Queues
 - Fax Machines
 - Settings
 - 3CX Phone System Updates
 - Links
 - Help

Edit VOIP Provider

Edit VOIP Provider settings and click OK or Apply to save changes

General | Advanced | Outbound Parameters | Inbound Parameters | Source ID | DID

Call Source Identification
The source of incoming calls must be identified. Configure how 3CX Phone System should identify calls from this provider.

Matching Strategy: Match All Fields

SIP Field: [] Variable: []

Add/Update Delete Clear Selection

SIP Field	Variable	Custom Value
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Source Identification by DID

If Call Source Identification is based on dialed number and DIDs are in use, you need to specify these DIDs here. Specify a Mask, or select individual DIDs

SIP Field containing DID numbers: Request Line URI: User Part

Source Identification by DID

2035955322

Add Mask Add DID Delete

OK Cancel Apply

Select did

Select all

<input type="checkbox"/> 2031234567
<input checked="" type="checkbox"/> 2035955322

OK Cancel