

# SIP Trunking using the Optimum Business SIP Trunk Adaptor and the 3CX IP-PBX

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Goal

The purpose of this configuration guide is to describe the steps needed to configure the 3CX IP-PBX for proper operation with Optimum Business SIP Trunking.

Prerequisites

Please follow the instructions in the Optimum Business SIP Trunk Set-up Guide. The Set-up Guide was left by the Optimum Business technician at installation. If you do not have the Set-up Guide, go to **optimumbusiness.com/sip** to download a copy. The guide describes the steps needed to configure the LAN side of the Optimum Business SIP Trunk Adaptor.

Important:

The Cablevision network only supports inband DTMF tones. In order for the 3CX IP-PBX to operate correctly with the Cablevision network, the Optimum SIP Trunk Adaptor must be enabled to convert out-of-band DTMF tones sent by the PBX to inband DTMF tones. To enable this conversion, log into the Optimum SIP Trunk Adaptor using the login and password specified in the Optimum SIP Trunk Adaptor Set-up Guide. On the **SIP Trunk Configuration** page, you **must** check the **Convert Inband DTMF** checkbox, and click the **Submit** button to update this setting. This is Step 3 of the Optimum Sip Trunk Set-up Guide.

This configuration guide provides the configuration steps for both PBX registration and static IP or non-registration modes of PBX operation.

PBX Information

Manufacturer:	3CX
Model:	Standard Edition
Software Version:	v12.0.32495.362
Does the PBX send SIP Registration messages (Yes/No)?	Yes

## 3CX PBX Configuration

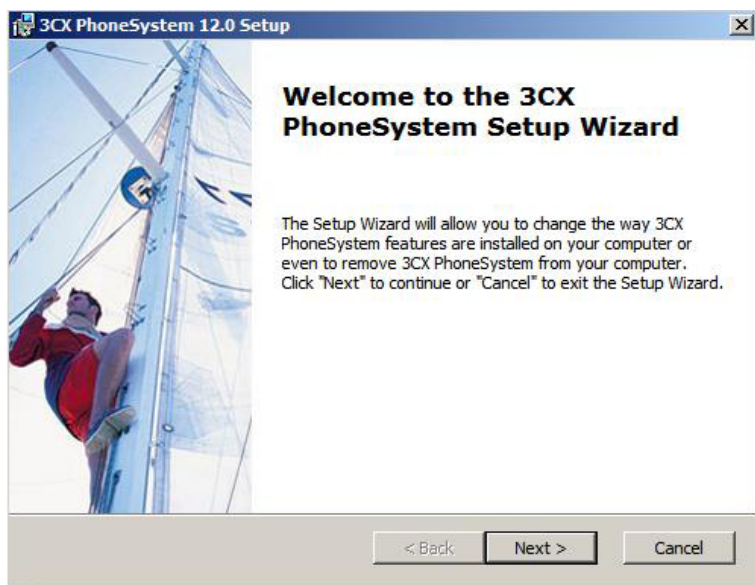
The steps below describe the basic configuration required to enable the 3CX PBX to use Optimum Business SIP Trunking for inbound and outbound calling. Please refer to the 3CX documentation for other advanced PBX features.

The configuration described here assumes that the PBX is already configured and operational with station side phones using assigned extensions or DIDs. This configuration is based on 3CX version 12.0.

## Basic Setup

The 3CX Phone System is a software-based VoIP IP-PBX for Microsoft Windows. In the lab, the PBX was set up by downloading the 3CX Phone System to a Windows 7 PC that comes with one Ethernet port. The PBX's Ethernet port, the local SIP phones, and Optimum Business Sip Trunk Adaptor's LAN port should be in the same LAN segment. The Optimum Business Sip Trunk Adaptor's LAN port and the PBX's Ethernet port have been assigned with IP addresses of 192.168.1.200 /24 and 192.168.1.155 /24 respectively.

Initially, the 3CX Phone System Setup Wizard must be completed first.



As soon as the PBX is downloaded, the 3CX Phone System Setup Wizard will start to install the PBX. From here basic settings such as LAN IP address, Administrator Login, adding extensions, and VOIP Gateway will be configured. Upon finishing, open the 3CX Windows Management Console that can be accessed from the Programs group. It will start by prompting a login. Enter login information and thereafter PBX configuration can begin.



## Creating Extensions

Navigate to **3CX Phone System ▶ Extensions ▶ 100** and click **100** to start setting up extension 100 for the phone. When creating new extensions, click **Add Extension** from top.

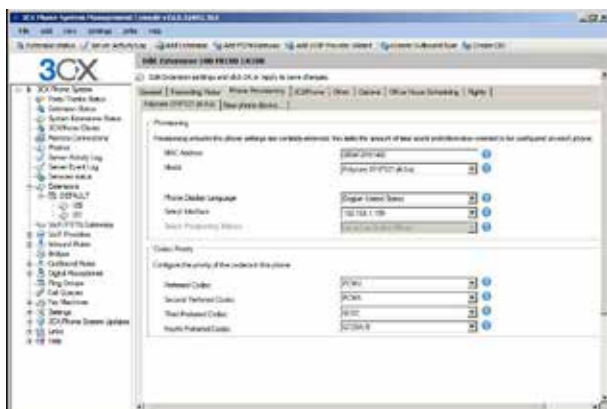
**Note:** the 3CX Phone System is implemented as Windows services. These Windows services are subject to firewall rules of the Windows PC (i.e.: firewall may block SIP registration packets from the SIP phones) and may be shut down due to certain default Windows settings. To prevent the firewall rules from blocking the desirable inbound traffic, certain firewall rules may need to be removed; to keep the PBX services running without being put to sleep, the setting of “Put the computer to sleep” should be set to “Never”.

Select the **General** tab. Enter the first name of the user in the **First Name** field and enter the last name of the user in the **Last Name** field. Enter a password in the **Password** field and leave other fields as default. When done click **OK** at the bottom of the page.

**Note:** the SIP phone assigned for this extension must use the authentication information from this page in order to successfully register with the PBX.



Select the **Phone Provisioning** tab, enter the SIP phone's MAC address in the **MAC Address** field. Select the phone's model from the drop-down list of the **Model** field and enter the PBX's IP address ("192.168.1.155" in this example) in the **Select Interface** field. Leave other fields as default and then click **OK** at the bottom of the page.



Select the **Other** tab and enter the DID assigned for the extension in the **Outbound Caller ID** field. Leave other fields as default and then click **OK** at the bottom of the page.

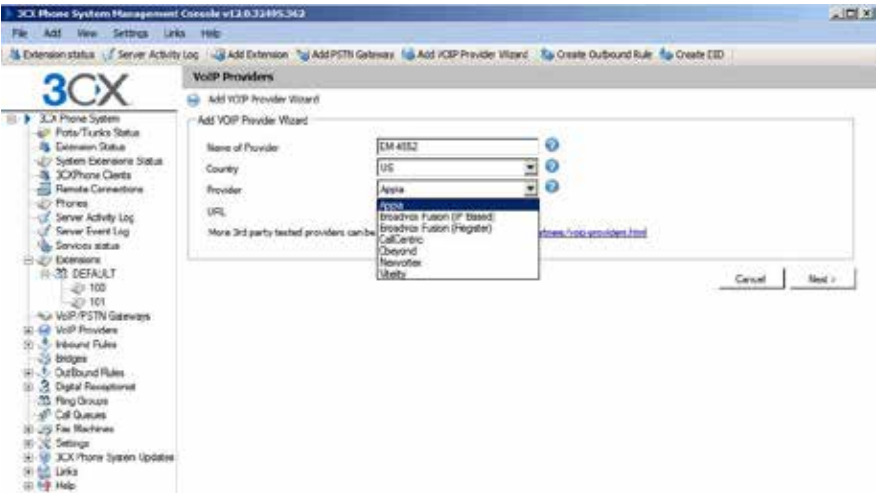


For additional extensions navigate to **Extensions** and then click **Add Extension**. Simply repeat above process for each new extension added.

## VoIP Provider Setup

Navigate to **VoIP Providers** to set up the Optimum Business SIP Trunk Adaptor as a VoIP provider for SIP registration mode. To add a new VoIP provider, click **Add Provider** under the VoIP Provider heading.

Enter a descriptive name in the **Name of Provider** field (“EM-4552” was used in this example), select the country and then the preferred Provider from menu. When done click **Next** at the bottom of the page.



Enter the Optimum Business SIP Trunk Adaptor’s IP address in both the **SIP server hostname or IP** field and the **Outbound proxy hostname or IP** field. This is the IP Address you entered in Step 2 of the Optimum Business SIP Trunk Set-up Guide. Enter **5060** in both the **SIP server port** and **Outbound proxy port** fields and then click **Next**.

### Provider Details

Enter the hostname and port of your provider’s SIP Server.

SIP server hostname or IP

192.168.1.200

SIP server port

5060

Outbound proxy hostname or IP

192.168.1.200

Outbound proxy port (default is 5060)

5060

Enter Pilot DID in the **External Number** field. Enter the authentication ID in the **Authentication ID** field and password in the **Authentication Password** field. Enter the maximum number of simultaneous calls in the **Maximum simultaneous calls** field and then click **Next**.

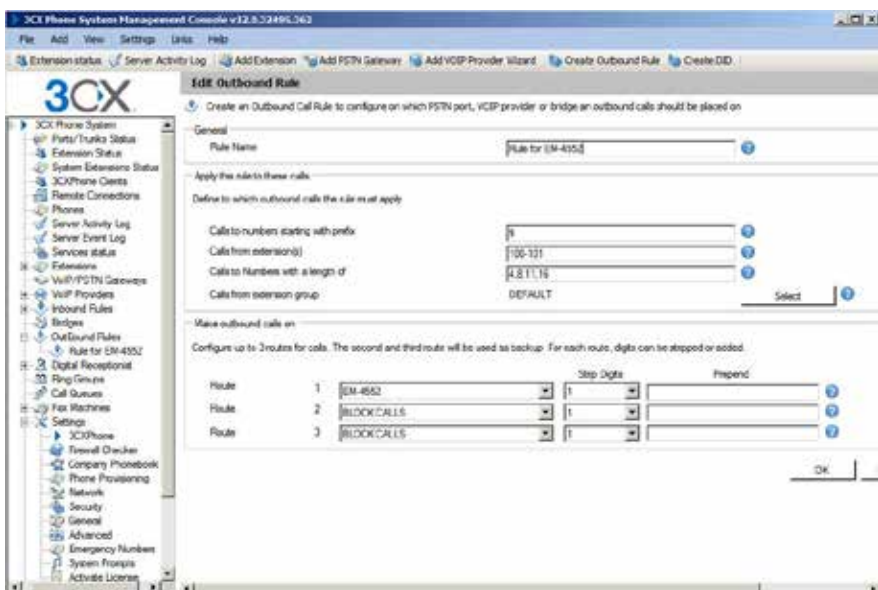
Account Details	
Enter the Authentication ID or SIP User, Password and number of your account	
External Number	<input type="text" value="4085551234"/> ?
Authentication ID	<input type="text" value="4085551234"/> ?
Authentication Password	<input type="password" value="*****"/> ?
3 Way Authentication ID	<input type="checkbox"/> <input type="text"/> ?
Simultaneous Calls	
Maximum Simultaneous Calls	<input type="text" value="4"/> ?



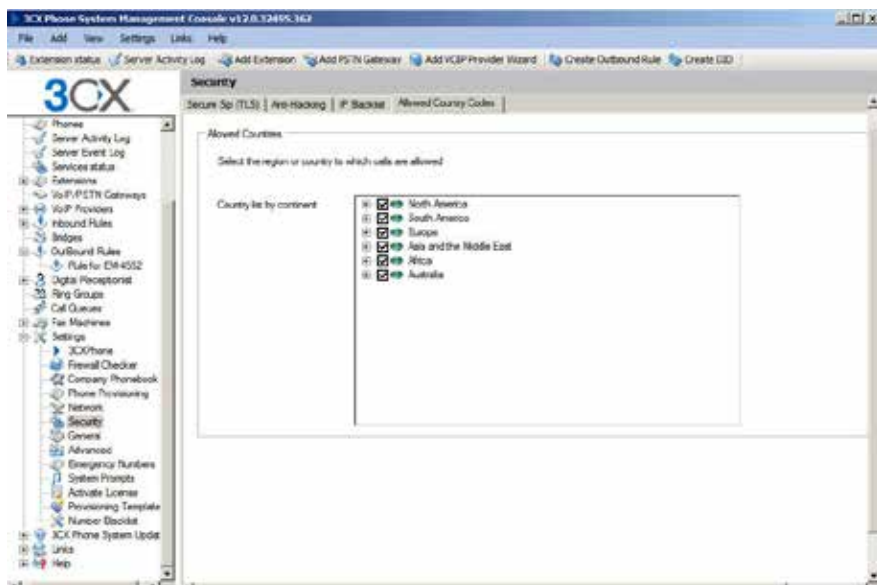
## Dial Plan

Navigate to **Outbound Rules** then click **Add Outbound Rule**. It must be given a name. Enter **9** in the **Calls to numbers starting with prefix** field (this allows dialing "9" first to access the SIP trunks). Enter **100-101** in the **Calls from extension(s)** field (this allows extension 100,101 in this example to access the SIP trunks). Enter **4,8,11,16** in the **Calls to Numbers with a length of** field (this allows "9+ 3-digit, 9+ 7-digit, 9+ 10-digit, 9+ 15-digit" dialing). Leave other fields as default and click **Finish**.

**Note:** For 911 calls a separate rule needs to be created and added above this rule otherwise the 9 will be stripped.



For International Calls the country code should be checked within included country codes. To verify this navigate to **Settings ▶ Security ▶ Allowed Country Codes**.



**Note:** The DTMF tone duration generated by the phones needs to be increased from the default value of 180ms–200ms to 600ms. The PBX does not have the capability to change the DTMF settings, the change must be done on the phones.

### Important:

The Cablevision network only supports inband DTMF tones. In order for the 3CX IP-PBX to operate correctly with the Cablevision network, the Optimum SIP Trunk Adaptor must be enabled to convert out-of-band DTMF tones sent by the PBX to inband DTMF tones. To enable this conversion, log into the Optimum SIP Trunk Adaptor using the login and password specified in the Optimum SIP Trunk Adaptor Set-up Guide. On the **SIP Trunk Configuration** page, you **must** check the **Convert Inband DTMF** checkbox, and click the **Submit** button to update this setting. This is Step 3 of the Optimum Sip Trunk Set-up Guide.

## SIP Registration

Navigate to **VoIP Providers ▶ EM-4552** and then click the **Advanced** tab to configure parameters for registration and codec priority. **SIP server hostname or IP** and **Outbound proxy hostname or IP** will again be the Optimum Business SIP Trunk Adaptor's IP address. This IP address was assigned to the Optimum Business SIP Trunk Adaptor in Step 2 of the Optimum Business SIP Trunk Set-up Guide. Below reflects previous set-up configuration under **General**.

General	Advanced	Outbound Parameters	Inbound Parameters	Source ID	DID
<b>Provider Details</b>					
Enter the hostname and port of your provider's SIP Server.					
SIP server hostname or IP		192.168.1.200		?	
SIP server port		5060		?	
Outbound proxy hostname or IP		192.168.1.200		?	
Outbound proxy port (default is 5060)		5060		?	
<b>Account Details</b>					
Enter the Authentication ID or SIP User, Password and number of your account					
External Number		4085551234		?	
Authentication ID		4085551234		?	
Authentication Password		*****		? ***	
3 Way Authentication ID		<input type="checkbox"/>		?	
<b>Simultaneous Calls</b>					
Maximum Simultaneous Calls		4		?	

For Provider Capabilities, check **Supports Re-Invite** and **Supports Replace**. Then enter time between registration attempts in the **Time between registration attempts field**. For DTMF capability be sure to leave **PBX Delivers Audio** unchecked. Tone duration may be modified through the phone. Select the radio button of **Internal** for the **Which IP to use in 'Contact' field for registration** field. Assign desired codec. Leave other fields as default and click **OK**.

General	Advanced	Outbound Parameters	Inbound Parameters	Source ID	DID																		
<b>Provider Capabilities</b> Configure Advanced options Supports Re-Invite <input checked="" type="checkbox"/> ? Supports Replace <input checked="" type="checkbox"/> ? PBX Delivers Audio <input type="checkbox"/> ? Switch on Secure RTP (SRTP) <input type="checkbox"/> ? Disable Video <input type="checkbox"/> ?																							
<b>Registration Settings</b> Configure Advanced options Time between registration attempts (in seconds) <input type="text" value="60"/> ? Require registration for: <input type="text" value="In and Outgoing Calls"/> ? Which IP to use in 'Contact' field for registration: <input type="radio"/> External(STUN resolved) ? <input checked="" type="radio"/> Internal ? <input type="radio"/> Specified IP <input type="text"/> ?																							
<b>Codec priorities</b> Specify which codecs to use and their priority <table border="1"> <thead> <tr> <th>Available Codecs</th> <th></th> <th>Assigned Codecs</th> <th></th> </tr> </thead> <tbody> <tr> <td>G.711 Alaw</td> <td rowspan="4">           Add &gt;            &lt; Remove         </td> <td>G.711 Ulaw</td> <td rowspan="4">           Up            Down         </td> </tr> <tr> <td>GSM-FR</td> <td></td> </tr> <tr> <td>Speex</td> <td></td> </tr> <tr> <td>ILBC</td> <td></td> </tr> <tr> <td>G.722</td> <td></td> <td></td> <td></td> </tr> </tbody> </table>						Available Codecs		Assigned Codecs		G.711 Alaw	Add > < Remove	G.711 Ulaw	Up Down	GSM-FR		Speex		ILBC		G.722			
Available Codecs		Assigned Codecs																					
G.711 Alaw	Add > < Remove	G.711 Ulaw	Up Down																				
GSM-FR																							
Speex																							
ILBC																							
G.722																							

## Static IP Mode

Navigate to **VoIP Providers ▶ EM-4552** and then click **Outbound Parameters** to control caller ID for outbound calls in static IP mode.

Select **From: User Part** from the drop-down list of the **SIP Field**.

In the corresponding variable select **“OutboundCallerId” Outbound caller Id taken from Extension**. Click the **Add/Update** button. Leave other fields as default and click **OK**.

The screenshot shows the 'Outbound Parameters' tab in the 3CX configuration interface. The 'SIP Field' dropdown is set to 'From: User Part' and the 'Variable' dropdown is set to '“OutboundCallerId” Outbound caller Id taken from Extension'. The 'Add/Update' button is highlighted. Below the dropdowns is a table with the following data:

SIP Field	Variable	Custom Value
To: Host Part	“GWIHostPart” gateway/provider host part	
From: Display Name	“OutboundCallerId” Outbound caller Id taken from Extension	
From: Host Part	“GWIHostPart” gateway/provider host part	
Remote Party ID: Calling Party: User Part	“OutboundCallerId” Outbound caller Id taken from Extension	
Remote Party ID: Calling Party: Host Part	“GWIHostPart” gateway/provider host part	
From: User Part	“OutboundCallerId” Outbound caller Id taken from Extension	
Remote Party ID: Calling Party: Display Name	“OutboundCallerId” Outbound caller Id taken from Extension	



Now navigate to **VoIP Providers ▶ EM-4552** and next to **Require registration for:** change to **Do not require** to deliver calls without authentication.

The screenshot shows the 'Registration Settings' section of the 'Outbound Parameters' tab. The 'Require registration for:' dropdown is set to 'Do not require'. The 'Require registration for:' dropdown is highlighted. Below the dropdowns is a table with the following data:


Require registration for:	Require registration for:
Do not require	
Extended SIP URI (testing)	
Internal	
Specified IP	

Verify Status

To verify registration status of the extensions navigate to **Extension Status** to check if the phones have successfully registered with the Optimum Business SIP Trunk Adaptor.

Extension Status						
<div><div>Disconnect Call</div><div>Show Filter</div></div>						
	Status	Extension	User Status	DND	Queues	Name
	Registered (idle)	100	Available	OFF	OUT	FN100 LN100
	Registered (idle)	101	Available	OFF	OUT	FN101 LN101

Once the PBX has set up the Optimum Business SIP Trunk Adaptor as its SIP Trunk service provider, it should be able to register successfully. To verify registration, navigate to **Ports/Trunks Status** to check if PBX has successfully registered with Optimum Business SIP Trunk Adaptor.

Ports/Trunks Status				
<div><div>Disconnect Call</div></div>				
	Status	Virtual Extension Number	Type	Name
	Registered (idle)	10000	Provider	EM-4552

DID Assignment

Navigate to **VoIP Providers ▶ EM-4552** and then click the **DID** tab to enter the DIDs assigned for the SIP trunk service. Enter each DID one at a time in the field next to the **?** icon and then click the **◀ Add** button. When done click **OK**.

The screenshot shows the 'DID' tab of the configuration page for provider 'EM-4552'. The 'DID Numbers' section contains a list of four DIDs: 4085551234, 4085555555, 4085555556, and 4085555557. To the right of the list is an input field with a question mark icon, and buttons for '< Add' and 'Remove >'. A note at the top states: 'Enter the DID/DDI number(s) linked to this provider. An inbound rule will be automatically created for each DID/DDI and needs to be modified to route calls to the appropriate extensions. You might also need to configure source identification by DID/DDI from the Source ID tab.'

Navigate to **VoIP Providers ▶ EM-4552** and click the **Source ID** tab. Check the **Source identification by DID** checkbox to specify that dialed numbers from incoming calls may be routed to the extensions. Click **Add DID**. From the pop-up box, check the **Select all** checkbox and then click **OK**. After the DIDs are added, leave other fields as default then click **OK**.

The screenshot shows the 'Source ID' tab. The 'Source identification by DID' checkbox is checked. Below it, a note says: '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'. The 'SIP Field containing DID numbers' dropdown is set to 'Request Line URI : User Part'. In the 'Source Identification by DID' section, the same four DIDs are listed. To the right are buttons for 'Add Mask', 'Add DID', and 'Delete'.

Navigate to **VoIP Providers ▶ EM-4552**. The PD will appear here and remaining DIDs will appear after it as sub tabs. Click on each DID, one at a time, to map to corresponding extension. By way of example, extension 100 will be assigned to 4085555555. Select the radio button next to **Connect to Extension** and choose extension 100. Leave other fields as default and then click **OK**.

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

DID/DDI number/mask  
4085555555

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

☒ EM-4552

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  
☐ Include holidays

100 FN100 LN100  
  
102 sally  
100 FN100 LN100  
  
email of extension 888  
Set up Specific Office Hours

☒ Apply the same routing logic Outside of office hours  
☐ Play Holiday Prompt on Public Holiday

OK Cancel Apply



## Call Forward

Navigate to **Extensions** and select the extension desired for Call Forwarding. Upon selecting the extension, click on **Forwarding Rules** on top and from here configure the forwarding preference according to the different statuses shown. For example if call forwarding is desired when away, click **Away** then click on **An external number or Skype ID** from either the internal or external option and enter desired forwarding number. When finished click **OK**.

**Edit Extension-100 FN100 LN100**

Available	Away	Out of Office	Available 2	Out of Office 2	Exceptions
-----------	------	---------------	-------------	-----------------	------------

Configure how calls should be re-directed when a user is away.

Forward internal calls to

Forward all calls to:

- ☐ Send call to my voice mail
- ☐ Send call to my mobile number
- ☐ Send call to
- ☒ An external number or Skype ID
- ☐ Rebound "" (Offer option to Confirm to accept)
- ☐ Disconnect the call
- ☐ Don't forward calls outside office hours (Send these to my voice mail)

9408777777

Forward external calls to

Forward all calls to:

- ☐ Send call to my voice mail
- ☐ Send call to my mobile number
- ☐ Send call to
- ☒ An external number or Skype ID
- ☐ Rebound "" (Offer option to Confirm to accept)
- ☐ Disconnect the call
- ☐ Don't forward calls outside office hours (Send these to my voice mail)

9408777777

☐ Using 302 diversion header

Make sure the Call Forwarding configuration corresponds to its current status. To change current status of the extension, from the **Edit Extension** section after clicking on **Extensions** click the **Other** tab and change **Current status** to desired preference.

Edit Extension-100 FN100 LN100

General
Forwarding Rules
Phone Provisioning
3CXPhone
Other
Options
Office Hours Scheduling
Rights

User Information

Configure user status and options

Current status

Away

?

Current Status

Logged Out

?

DND

OFF

?

Outbound Caller ID

4085555555

?

SIP ID

?

Extension Capabilities

Some of the options below are enabled to overcome compatibility issues with either old phones or those not supporting all of the SIP features

PEX Delivers Audio

☐

?

Supports Re-Invite

☒

?

Support 'Replaces' header

☒

?

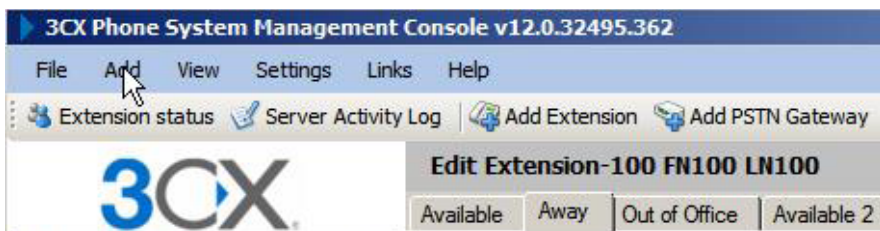
Switch on Secure RTP (SRTP)

☐

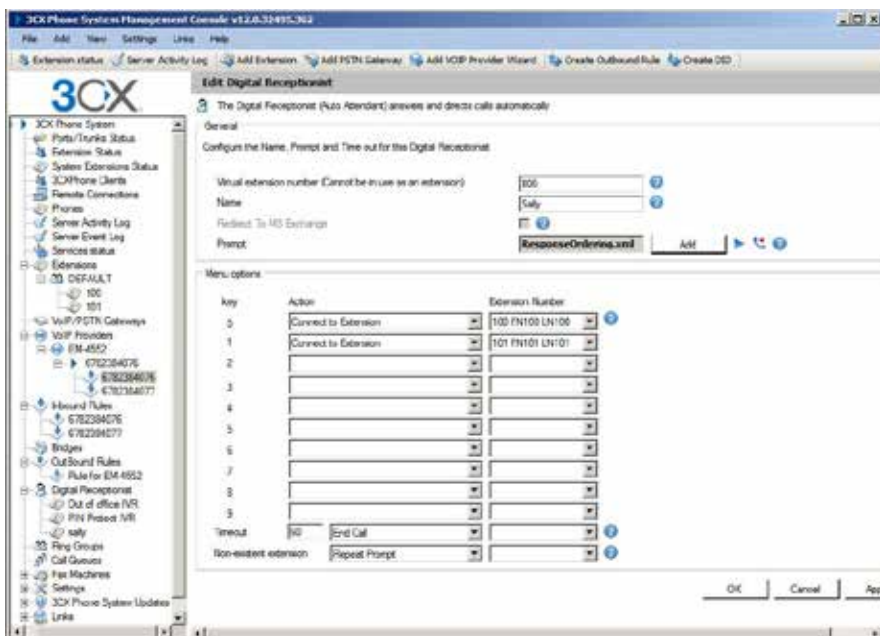
?

## Auto-Attendant

From **Add** button above select **Digital Receptionist**.



Upon clicking **Digital Receptionist**, give the auto-attendant an extension, name, and select the desired prompt. Under **Menu options** select how the auto-attendant should behave in accordance with each key. For example both 0 and 1 below direct the auto-attendant to extensions 100 and 101. When finished click **OK**.



Thereafter navigate to **VoIP Providers ▶ EM-4552**. Click the DID desired for auto-attendant then click on radio button of **Connect to Digital Receptionist**. Be sure to link it to extension of AA (in this example 102). When finished click **OK**.

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

4085555557

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

☒ EM-4552

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

☐ Include holidays

101 FN101 LN101

102 ealy

100 FN100 LN100

email of extension 000

Set up Specific Office Hours

☒ Apply the same routing logic Outside of office hours

☐ Play Holiday Prompt on Public Holiday

OK

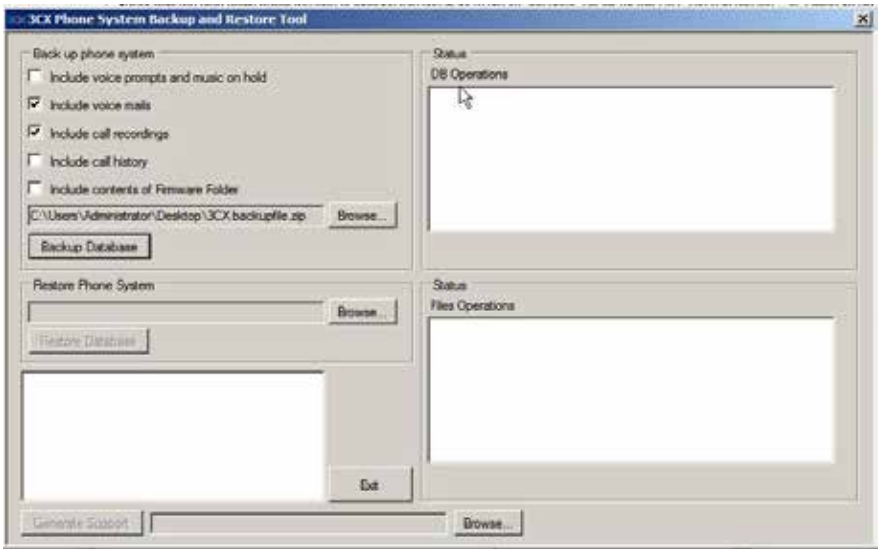
Cancel

Apply

## Backup/Restore

To Backup/Restore, from Program Groups select **Backup and Restore Tool**.





Select the desired preferences and when done click **Backup Database**. To restore click **Restore Database** just below.