

# SIP Trunking using the Optimum Business Sip Trunk Adaptor and the 3CX PBX v12.5

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## 1 Overview

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

## 2 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](http://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.

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

### PBX Information

<b>Manufacturer:</b>	3CX
<b>Model:</b>	Standard Edition
<b>Software Version:</b>	v12.5
<b>Does the PBX send SIP Registration messages (Yes/No)?</b>	Yes
<b>Vendor Contact:</b>	<a href="http://www.3cx.com">www.3cx.com</a>

## 3 PBX Configuration

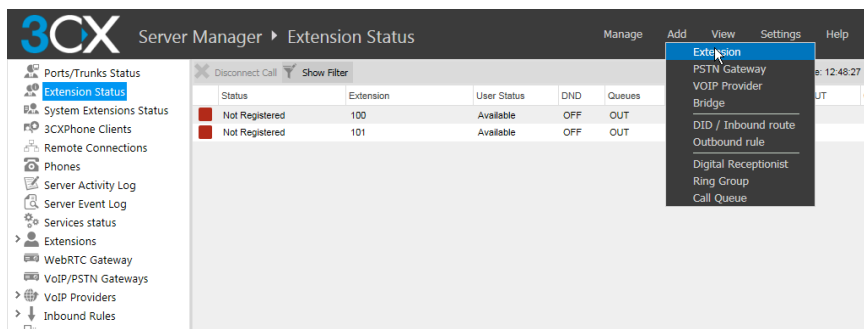
The steps below describe the minimum configuration required to enable the PBX to use Optimum Business Sip Trunking for inbound and outbound calling. Please refer to the 3CX product documentation for more information on advanced PBX features.

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 PBX version 12.5.

The 3CX Phone System is a software-based VoIP IP-PBX for Microsoft Windows. In this example, the PBX was set up by downloading the 3CX Phone System to a Windows7 PC which comes with one Ethernet port. The Optimum Sip Trunk Adaptor's (EdgeMarc) 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.

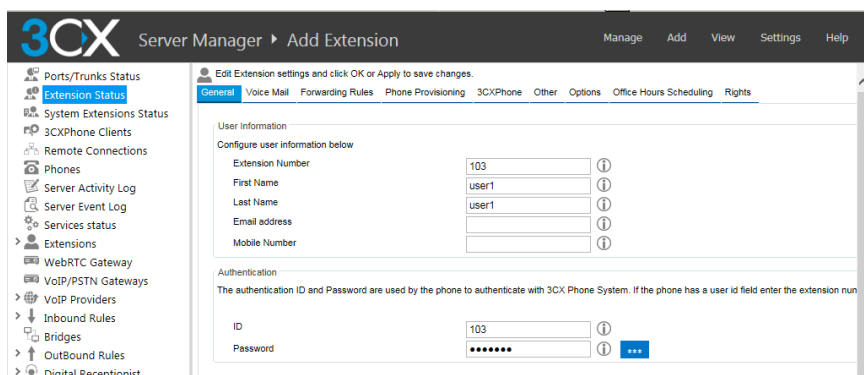
## 4 Creating Extensions

Start configuring the PBX by navigating to “**3CX Phone System**” → “**Extensions**” → “**100**” and click “**100**” to start setting up extension 100 for the phone. When creating new extensions, “**Add Extension**” must be clicked from above.



- a) Select the “**General**” tab, enter the first name of the user in the “**First Name**” field, enter the last name of the user in the “**Last Name**” field, enter “**admin**” in the “**Password**” field, leave other fields as default and click the “**OK**” button at the bottom of the page.

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



- b) 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, 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 the “**OK**” button at the bottom of the page.

**NOTE:** The PBX will only accept SIP registration for extension 100 from the phone with the specified MAC address.

The screenshot shows the '3CX Server Manager' interface with the 'Add Extension' page. The 'Phone Provisioning' tab is selected. The left sidebar contains a tree view with categories like Ports/Trunks Status, Extension Status, System Extensions Status, 3CXPhone Clients, Remote Connections, Phones, Server Activity Log, Server Event Log, Services status, Extensions, WebRTC Gateway, VoIP/PSSTN Gateways, VoIP Providers, Inbound Rules, Bridges, Outbound Rules, Digital Receptionist, Ring Groups, Call Queues, Fax Machines, and Settings. The main content area has a top bar with 'Edit Extension settings and click OK or Apply to save changes.' and tabs for General, Voice Mail, Forwarding Rules, Phone Provisioning (active), 3CXPhone, Other, Options, Office Hours Scheduling, and Rights. Below the tabs is a 'New phone device...' button. The 'Provisioning' section contains fields for MAC Address (3333 3333 3333), Model (Polycom SPP330 (3.3.x)), Phone Display Language (English United States), Time Zone (Use global PBX settings), Select Provisioning Method (Local Lan (in the Office)), and Select Interface (192.168.1.155). The 'Codes Priority' section lists Preferred Code (PCMU), Second Preferred Code (PCMA), Third Preferred Code (G729AB), and Fourth Preferred Code (Not Available).

c) Select the **“Other”** tab, enter the DID (pilot DID for the first extension in this example) assigned for the extension in the **“Outbound Caller ID”** field, leave other fields as default and then click the **“OK”** button at the bottom of the page.

**NOTE:** This outbound caller ID will be used only when the PBX is configured for static IP mode. For SIP registration mode, the pilot DID will be used as Caller ID.

The screenshot shows the '3CX Server Manager' interface with the 'Add Extension' page. The 'Other' tab is selected. The left sidebar is the same as in the previous screenshot. The main content area has a top bar with 'Edit Extension settings and click OK or Apply to save changes.' and tabs for General, Voice Mail, Forwarding Rules, Phone Provisioning, 3CXPhone, Other (active), Options, Office Hours Scheduling, and Rights. Below the tabs is a 'User Information' section with a 'Configure user status and options' button. The 'User Information' section contains fields for Current status (Available), Queues Status (Logged Out), DND (OFF), Outbound Caller ID (4085555556), and SIP ID. The 'Extension Capabilities' section lists options like PBX Delivers Audio, Supports Re-Invite, Support 'Replaces' header, Switch on Secure RTP (SRTP), and Enable Secure SIP (TLS). A warning message states: 'Secure SIP must be pre-configured from Security node > Secure SIP (TLS). Currently Secure SIP works only for 3CXPhone for Windows and MAC Clients.'

## 5 VoIP Provider Setup

Navigate to **“VOIP Providers”** to set up the Optimum Sip Trunk Adaptor (Edgemark) as a VoIP provider for SIP registration mode. To add a new VoIP provider, click the **“Add Provider”** link under the VOIP Provider heading.

Enter a descriptive name in the **“Name of Provider”** field, **EM-4552** was used in this example. Select the US from the **“Country”** drop down and **Optimum Business** from the **“Provider”** drop down menu. When done click the **“Next”** button at the bottom of the page.

**3CX** Server Manager ▸ Add VOIP Provider

Manage Add View Settings Help

Ports/Trunks Status  
Extension Status  
System Extensions Status  
3CXPhone Clients  
Remote Connections  
Phones  
Server Activity Log  
Server Event Log  
Services status  
Extensions  
WebRTC Gateway  
VoIP/PSTN Gateways  
VoIP Providers  
Inbound Rules

Add VOIP Provider Wizard

Add VOIP Provider Wizard

Name of Provider: EM-4552 ⓘ

Country: US ⓘ

Provider: Optimum Business ⓘ

URL: <http://www.optimumbusiness.com/business-phone/sip-trunking.jsp>

3CX Supported VoIP Providers can be found here: <http://www.3cx.com/partners/sip-trunks/>

More 3rd party tested providers can be found here: <http://www.3cx.com/partners/voip-providers.html>

Cancel Next >

- a) Enter the Optimum Business Sip Trunk Adaptor's (Edgemark'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”** field and the **“Outbound proxy port”** (default is 5060) field and then click the **“Next”** button.

**3CX** Server Manager ▸ Add VOIP Provider

Manage Add View Settings Help

Ports/Trunks Status  
Extension Status  
System Extensions Status  
3CXPhone Clients  
Remote Connections  
Phones  
Server Activity Log  
Server Event Log  
Services status  
Extensions  
WebRTC Gateway  
VoIP/PSTN Gateways  
VoIP Providers  
Inbound Rules

Add VOIP Provider Wizard

VOIP Provider Details:

Enter the hostname and port for your VOIP 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 ⓘ

< Back Next >

- b) Enter the Pilot DID in the **“External Number”** field, enter the authentication ID (if needed) in the **“Authentication ID”** field, enter the authentication password in the **“Authentication Password”** field, enter the maximum number of simultaneous calls in the **“Maximum simultaneous calls”** field and then click the **“Next”** button.

**3CX** Server Manager ▸ Add VOIP Provider

Manage Add View Settings Help

Ports/Trunks Status  
Extension Status  
System Extensions Status  
3CXPhone Clients  
Remote Connections  
Phones  
Server Activity Log  
Server Event Log  
Services status  
Extensions  
WebRTC Gateway  
VOIP/PSTN Gateways  
VOIP Providers

Add VOIP Provider Wizard

Account Details  
Enter the Authentication ID or SIP User, Password and number of your account

External Number 4085555556 ⓘ  
Authentication ID (aka SIP User ID) 4085555556 ⓘ  
Authentication Password ..... ⓘ  
3 Way Authentication ID ☐ ⓘ

Simultaneous Calls  
Maximum simultaneous calls 4 x ⓘ

< Back Next >

- c) Leave all fields as default and then click the **“Next”** button.

Add VOIP Provider Wizard

Office Hours  
Configure where calls should be routed during office hours.

☐ End Call  
☐ Connect to Extension 100 FN100 LN100 ⓘ  
☐ Connect to Queue / Ring Group ⓘ  
☒ Connect to Digital Receptionist 102 sally ⓘ  
☐ Voicemail box for Extension 100 FN100 LN100 ⓘ  
☐ Forward to Outside Number ⓘ  
☐ Send fax to email of extension email of extension 888 ⓘ

☒ Same as Out of Office hours

< Back Next >

## 6 Dial Plan / Outbound Rules

Navigate to the **“Edit Outbound Rule”** section. If not here yet, click on **“Outbound Rules”** from menu then **“Rule for EM-4552”**. 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 then click the **“Finish”** button. Alternatively, type **“0-16”** & that will include all calls.

3CX Server Manager ▸ Edit Outbound Rule - Rule for EM-4552... Manage Add View Settings Help

Create an Outbound Call Rule to configure on which PSTN port, VOIP provider or bridge an outbound calls should be placed on

**General**

Rule Name: Rule for EM-4552

Apply this rule to these calls

Define to which outbound calls the rule must apply

Calls to numbers starting with prefix		?
Calls from extension(s)	100-103	?
Calls to Numbers with a length of	1,2,3,4,7,8,10,11,15	?
Calls from extension group	DEFAULT	

Select

Make outbound calls on

Configure up to 3 routes for calls. The second and third route will be used as backup. For each route, digits can be stripped or added.

Route		Strip Digits	Prepend
Route 1	EM-4552	0	
Route 2	BLOCK CALLS	1	
Route 3	BLOCK CALLS	1	


OK Cancel Apply

For International Calls, make sure the country code is checked within included country codes. To confirm this, Navigate to **“Settings”→“Security”→“Allowed Country Codes”**.





The screenshot shows the 'Security Settings' window with the 'Allowed Country Codes' tab selected. The 'Allowed Countries' section contains the instruction 'Select the region or country to which calls are allowed'. Below this, a list titled 'Country list by continent' shows six entries, each with a globe icon and a checked checkbox: North America, South America, Europe, Asia and the Middle East, Africa, and Australia.

Security Settings			
Secure SIP (TLS)	Anti-Hacking	IP Blacklist	Allowed Country Codes
<b>Allowed Countries</b>			
Select the region or country to which calls are allowed			
<b>Country list by continent</b>			
>	<input checked="" type="checkbox"/>		North America
>	<input checked="" type="checkbox"/>		South America
>	<input checked="" type="checkbox"/>		Europe
>	<input checked="" type="checkbox"/>		Asia and the Middle East
>	<input checked="" type="checkbox"/>		Africa
>	<input checked="" type="checkbox"/>		Australia

**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.

## 7 Registration Mode Parameters

Navigate to “**VOIP Providers → EM-4552**” and under General is where Authentication credentials should be entered. The Authentication credentials must match what is configured in the Optimum Business Sip Trunk Adaptor. This is step 3 of the Optimum Business Sip Trunk Set up Guide.

The screenshot shows a web form for configuring SIP registration parameters. It is divided into two main sections: 'Account Details' and 'Simultaneous Calls'. The 'Account Details' section has a header 'Enter the Authentication ID or SIP User, Password and number of your account'. It contains four input fields: 'External Number' (with value 4085555555), 'Authentication ID' (with value 4085555555 and a clear button 'x'), 'Authentication Password' (masked with dots), and '3 Way Authentication ID' (with an unchecked checkbox). Each field has an information icon. The 'Simultaneous Calls' section has a single input field 'Maximum Simultaneous Calls' with the value 4 and an information icon.

Account Details	
Enter the Authentication ID or SIP User, Password and number of your account	
External Number	4085555555 ⓘ
Authentication ID	4085555555 x ⓘ
Authentication Password	..... ⓘ ***
3 Way Authentication ID	<input type="checkbox"/> ⓘ

Simultaneous Calls	
Maximum Simultaneous Calls	4 ⓘ

Thereafter click the “**Advanced**” tab to configure parameters for registration and codec priority. **SIP server hostname & Outbound proxy hostname** will be the IP address assigned to the Optimum Business Sip Trunk Adaptor. This was step 2 in the Optimum Business Sip Trunk Set up Guide.

- For Provider Capabilities, check “**Supports Re-Invite**” & “**Supports Replace**”. Then enter time between registration attempts in the “**Time between registration attempts (in seconds)**” field.
- Select the radio button for “**Local IP Address**” for the “**Which IP to use in ‘Contact’ field for registration**” field.
- Assign Codec G.711 U-law
- Leave other fields as default and then click the “**OK**” button.

**3CX Server Manager** ▶ Edit VOIP Provider - EM-4552

Manage Add View Settings Help

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

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

**Provider Capabilities**  
Configure Advanced options

- Supports Re-invite ☒ ⓘ
- Supports 'Replace' ☒ ⓘ
- PBX Delivers Audio ☐ ⓘ
- Switch on Secure RTP (SRTP) ☐ ⓘ
- Disable Video ☐ ⓘ

**Registration Settings**  
Configure Advanced options

Time between registration attempts (in seconds)  ⓘ

Require registration for:  ⓘ

Which IP to use in 'Contact' and 'Connection' SIP fields

- ☐ Use Default Settings ⓘ
- ☒ Local IP Address (Windows Default Route) ⓘ
- ☐ Use this IP Address  ⓘ

**Codec priorities**  
Specify which codecs to use and their priority

Available Codes	Assigned Codes
G.711 A-law	G.711 U-law
GSM-FR	
Speex	
iLBC	
G722	

Buttons: Add > < Remove Up Down

**Important:** For both Registration and Static IP mode of operation

**DTMF Configuration:** Leave **“PBX Delivers Audio”** unchecked. This is important for outgoing DTMF calls to work. The Cablevision network only supports in-band DTMF tones.

### DTMF Tone Duration:

The DTMF tone duration must be modified on each phone. Some phones have a default setting between 180ms and 200ms. This setting is too low. The recommended setting is 600ms.

## 8 Static IP Mode Parameters

Navigate to **“VOIP Providers”** → **“EM-4552”** and then click the **“Outbound Parameters”** to control caller ID for outbound calls in static IP mode.

**NOTE:** This section is needed for static IP mode only. Also note that PBX would be considered “registered” (trusted) without the SIP registration process in static IP mode.

- a) Select **“From: User Part”** from the drop-down list of the **“SIP Field”** and have it correspond to **“Outbound caller Id taken from Extension”** from the **“Variable”** list.

3CX Server Manager ▶ Edit VOIP Provider - EM-4552

Manage Add View Settings Help

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

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

Assign 3CX call variables to SIP header fields  
Configure which SIP message fields should contain what information. Requires SIP knowledge - misconfiguration will cause your PBX to malfunction.

SIP Field	Variable	Custom Value
Request Line URI : User Part	"CalledNum" number that has been dialed (default: To-> v	
Request Line URI : Host Part	"GWHostPort" gateway/provider host/port	
Contact : User Part	"AuthID" authentication	
Contact : Host Part	"ContactUri" usually, content of Contact field	
To : Display Name	"CalledName" name that has been dialed (default: To-> v	
To : User Part	"CalledNum" number that has been dialed (default: To-> v	
To : Host Part	"GWHostPort" gateway/provider host/port	
From : Display Name	"OutboundCallerId" Outbound caller Id taken from Exten	
From : User Part	"OutboundCallerId" Outbound caller Id taken from Exten	
From : Host Part	"GWHostPort" gateway/provider host/port	
User Agent : Text String	Leave default value	
Remote Party ID - Called Party : Display Name	Leave default value	
Remote Party ID - Called Party : User Part	Leave default value	
Remote Party ID - Called Party : Host Part	Leave default value	

- b) Leave other fields as default and click **“Apply”**.

Also when configuring Static mode, navigate to **“VoIP Providers”** → **“EM-4552”** and then next to **“Require registration for:”** change the Registration to **“Do not require”** to deliver calls without authentication.

Registration Settings  
Configure Advanced options

Time between registration attempts (in seconds)  ⓘ

Require registration for:  ⓘ

Which IP to use in 'Contact' and 'Connection' SIP fields

☐ Use Default Settings ⓘ  
☒ Local IP Address (Windows Default Route) ⓘ  
☐ Use this IP Address  ⓘ

## 9 Assigning DIDs

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**”.

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

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

**DID Numbers**

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

4085555556  
4085555557

?

< Add

Remove >

Navigate to “**VOIP Providers**” → “**EM-4552**” and then click the “**Source ID**” tab and then check the “**Source identification by DID**” checkbox to specify which 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 the “**OK**” button.
- After the DIDs are added, leave other fields as default and then click the “**OK**” button.

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

☒ **Source identification by DID**

If Call Source identification is based on dialled 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 To : User Part

Source Identification by DID

4085555556  
4085555557

Add Mask  
Add DID  
Delete

Navigate to **“VOIP Providers” → “EM-4552”**. Find the trunk and then click on each DID (one at a time) to map to its assigned extension. In this example, extension 100 is assigned to “4085555556”.

- Select the radio button next to **“Connect to Extension”**.
- Select the extension from the extension drop-down list.
- Leave other fields as default and then click the **“OK”** button

## 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

4085555556



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

Available ports

&gt; EM-4552



## Office Hours

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

☐ End Call☒ Connect to Extension

100 FN100 LN100

☐ Connect to Queue / Ring Group☐ Connect to Digital Receptionist

102 sally

☐ Voicemail box for Extension

100 FN100 LN100

☐ Forward to Outside Number☐ Send fax to

email of extension 888


☐ Set up Specific Office Hours

Set up Specific Office Hours

☐ Include holidays

## 10 Call Forward

From main menu, navigate to “**Extensions**” and select the extension 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 to configure call forwarding when away, click on “**Away**” and 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 settings and click OK or Apply to save changes.

General	Voice Mail	<b>Forwarding Rules</b>	Phone Provisioning	3CXPhone	Other	Options	Office Hours Scheduling	Rights
Available	<b>Away</b>	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)

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) ☐ Using 302 diversion header  
☐ Disconnect the call  
☐ Don't forward calls outside office hours (Send these to my voice mail)

Make sure that 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**”, go to “**Other**” and change “**Current status**” to preference.



Edit Extension settings and click OK or Apply to save changes.

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

User Information

Configure user status and options

Current status	Away	i
Queues Status	Logged Out	i
DND	OFF	i
Outbound Caller ID	4085555556	i
SIP ID		i

Extension Capabilities

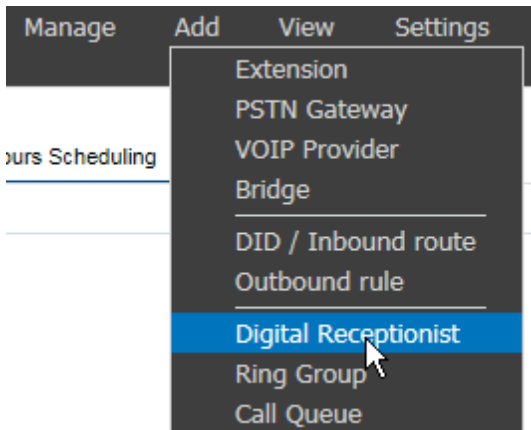
Some of the options below are enabled to overcome compatibility issues with either old phones or those not supporting all of the SIP features

PBX Delivers Audio	<input type="checkbox"/>	i
Supports Re-Invite	<input checked="" type="checkbox"/>	i
Support 'Replaces' header	<input checked="" type="checkbox"/>	i
Switch on Secure RTP (SRTP)	<input type="checkbox"/>	i
Enable Secure SIP (TLS)	<input type="checkbox"/>	i


⚠ Secure SIP must be pre-configured from Security node > Secure SIP (TLS).  
Currently Secure SIP works only for 3CXPhone for Windows and MAC Clients

## 11 Auto-attendant

From “**Add**” button on very top 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 & 1 below direct the auto-attendant to extensions 100 & 101. When finished click “**OK**”.


The Digital Receptionist (Auto Attendant) answers and directs calls automatically

General

Configure the Name, Prompt and Time out for this Digital Receptionist


Virtual extension number (Cannot be in use as an extension)

800




Name

Sally



Redirect To MS Exchange


☐





Prompt


ResponseOrdering.xml

Add












Menu options

key	Action	Extension Number	
0	Connect to Extension	100 FN100 LN100	
1	Connect to Extension	101 FN101 LN101	
2			
3			
4			
5			
6			
7			
8			
9			
Timeout	60	End Call	
Non-existent extension	Repeat Prompt		

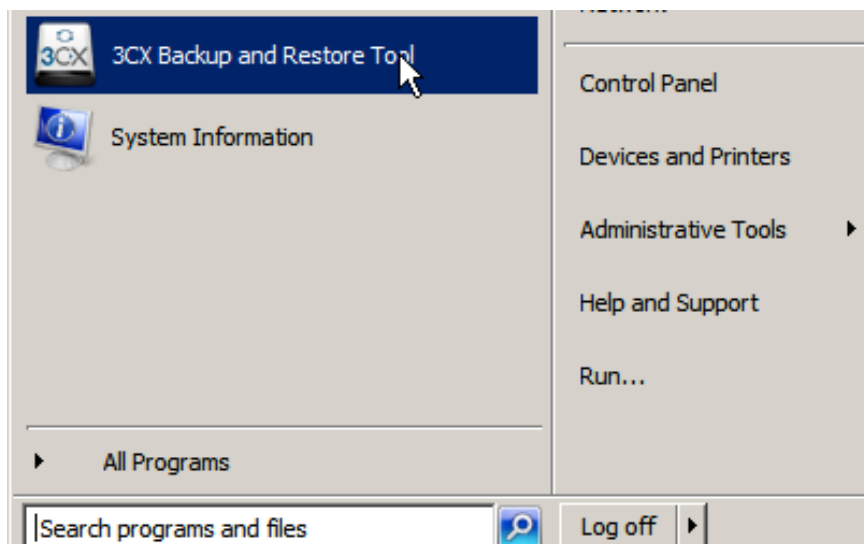
OK

Cancel

Apply

## 12 Backup/Restore

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



The tool will do a walkthrough for either a backup or a restore of the configuration file.