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 Windows7 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.

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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.

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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.

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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.

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



VolP 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	0
SIP server port	5060	0
Outbound proxy hostname or IP	192.168.1.200	0
Outbound proxy port (default is 5060)	5060	0

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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	4085551234	
Authentication ID	4085551234	
Authentication Password		0
3 Way Authentication ID		0
Simultaneous Calls		
Maximum Simultaneous Calls	4	0



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.

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

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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.

Prov	vider Details -				
Ente	r the hostnam	e and port of your provide	er's SIP Server.		
SIP	server hostnar	ne or IP		192.168.1.200	0
SIP :	server port			5060	0
Outb	ound proxy ho	ostname or IP		192.168.1.200	0
Outb	ound proxy po	ort (default is 5060)		5060	0
Ente	r the Authentio	cation ID or SIP User, Pa	assword and number of	your account	
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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**.

Provider Capabilities			
Configure Advanced options			
Supports Re-Invite	S		
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PBX Delivers Audio	E 🕢		
Switch on Secure RTP (SRTP)	E 🕢		
Disable Video			
Registration Settings			
Configure Advanced options			
Time between registration attempts (in secon	ds)	60 🕜	
Require registration for:		in and Outgoing Calls 📃 🔮	
Which IP to use in 'Contact' field for registrat	ion:	C Edemal(STUN resolved)	0
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		C Specified IP	0
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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.

Location of the desiredon number Specify in which SIP number field the dated	summer will be rough	the the stanges are required if you are using a supported gala	ning (srevider	
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Now navigate to **VoIP Providers > EM-4552** and next to **Require registration for**: change to **Do not require** to deliver calls without authentication.

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Configure Advanced options					
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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 Adapor.

Exte	nsion Status					
S Dis	connect Call 🛛 🖄 Show Fil	ter				
	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.

Port	s/Trunks Status			
S& Dis	sconnect Call			
	Status	Virtual Extension Number	Туре	Name
	Registered (idle)	10000	Provider	EM-4552

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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**.

Enter the DID/DDI nur	steria) Inked to this provider. An inbound rule will be automatically created for each DID/DDI and needs to be modified to noute calls to
appropriate extensions	Tou regist also need to configue source identification by UK/UU monithe source iD tab
4085555555 4085555556	
4085555557	5.00

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.

Source identification by DID		
If Call Source identification is based	on dialled number and DIDs are in use, you need to spe	ofy these DIDs here. Specify a Mask, or select individual DIDs
SIP Field containing DID numbers	Request Line URI : User Part	20
Source Identification by DID		
4085551234	Add Mask	
4085555556 4085555557	Add DID	
	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**.

ribound huie type	DID/DDI number/mask	0
DID/DOI number/mask:	4085505555	0
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elect the Gateway you want this DID/DDI rule to be dividual ports.	applied to. You can select on the whole gateway which	will apply the rule to all the ports, or you can select
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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**.

valiable	Away	Out of Office	Available 2	Out of Office 2	Exceptions	1	
Configu	re how ca	alls should be re-	-directed when	a user is away.			
Forward	d internal	calls to					
Forward	all calls	to:					
C Ser	d call to	my voice mail					
O Ser	nd call to	my mobile numb	er				_
C Ser	nd call to						• 0
⊙ An	external r	number or Skype	ID		94087777777		0
🗆 Ret	oound "	(Offer option to (Confirm to acce	ept)			
C Disc	connect t	he call					
Dor	nt forward	d calls outside of	ffice hours (Sen	nd these to my void	ce mail)		
Forward	n't forward d external	d calls outside of I calls to	ffice hours (Sen	nd these to my void	ce mail)		
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Forward Forward O Ser	n't forward d external d all calls t nd call to	d calls outside of l calls to to: my voice mail	ffice hours (Sen	nd these to <mark>my voi</mark> d	ce mail)		
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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		
ieneral Forwarding Rules Phone Provisioning	3CXPhone Other Options Office Hours Scheduling Rights	
User Information		
Configure user status and options		
Current status	Away 🔽 🥝	
Queues Status	Logged Out	
DND	off 💽 😒	
Outbound Caller ID	4085555555 📀	
SIP ID	0	
Extension Capabilities		
Some of the options below are enabled to overcon	ne compatibility issues with either old phones or those not supporting all of the	he SIP features
PBX Delivers Audio		
Supports Re-Invite	() 되	
Support 'Replaces' header	() ସ	
Sandrah um Sannam DTR /SDTR	E 0	



Auto-Attendant

From Add button above select Digital Receptionist.

🕨 3CX	Phone	System	n Manager	ment Co	nsole v1	2.0.324	95.362	
File	Add	View	Settings	Links	Help			
🕴 🖏 Ex	tension s	tatus 🔨	💰 Server A	ctivity Lo	g 🕼 🖓 A	dd Exten	sion 🙀 Add P	STN Gateway
	9				Edit Ext	ension	-100 FN100	LN100
	5	C,	Χ.	A	wailable	Away	Out of Office	Available 2

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**.

OCV	Edit Digital F	leoptionist						
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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**.

Inbound Rule type	DID/DDI nunber/mask	10
DID/DDI number/mask	4085555557	0
Apply this rule to these ports		
Select the Gateway you want this DID/DDI rule to be apple select individual pots.	ed to. You can select on the whole gateway which	will apply the rule to all the ports, or you can
Available ports	® 🖬 🖬 EM-4852	0
Office Hours		
Contigure where calls to this D1D/DD1 should be routed dur	ing office hours.	
C End Call	F	
C Connect to Extension	101 FN101 LN101	
C Connect to Queue / Ring Group	k.	- 0
	102 safy	- O
Connect to Digital Receptionist		
Connect to Digital Receptioniat Voicemail box for Extension	100 FN100 LN100	<u> </u>
Connect to Digital Receptioniat Voicemail box for Extension Forward to Outside Number	100 FN100 LN100	20
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optimum.

Backup/Restore

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





Back up phone system	2 Salue	
Include voice prompts and music on hold	DB Operations	
7 Include voice mails	R	
Vinclude call recordings		
Include call history		
include contents of Fernware Folder		
2/Users/Administrator/Desktop/3CX backupfile.zp Brows	e	
Rickup Database		
Restore Phone System	Satur	
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Select the desired preferences and when done click **Backup Database**. To restore click **Restore Database** just below.