



breakfree
with a software based PBX for Windows



Manual

3CX Phone System for Windows

Version 8.0

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1. Introduction to 3CX Phone System for Windows

What is 3CX Phone System for Windows?

3CX Phone System is a software-based IP PBX that replaces a traditional PBX and delivers employees the ability to make, receive and transfer calls. The IP PBX supports all traditional PBX features. An IP PBX is also referred to as a VOIP Phone System, IP PABX or SIP server.

Calls are sent as data packets over the computer data network instead of via the traditional phone network. Phones share the network with computers and separate phone wiring can therefore be eliminated.

With the use of a VOIP gateway, you can connect existing phone lines to the IP PBX and make and receive phone calls via a regular PSTN line. The 3CX phone system uses standard SIP software or hardware phones, and provides internal call switching, as well as outbound or inbound calling via the standard phone network or via a VOIP service.

Benefits of an IP Phone System / IP PBX

Much easier to install & configure than a proprietary phone system:

A software program running on a computer can take advantage of the advanced processing power of the computer and user interface of Windows. Anyone with an understanding of computer networks and windows can install and configure the PBX. A proprietary phone system often requires an installer trained on that particular proprietary phone system.

Easier to manage because of web based configuration interface:

A VOIP phone system has a web based configuration interface, allowing you to easily maintain and fine tune your phone system. Proprietary phone systems often have difficult to use interfaces which are designed so that only the phone system installers can use it effectively.

Call cost reduction:

You can save substantially by using a VOIP service provider for long distance or international calls. Easily connect phone systems between offices/branches via the Internet or WAN and make free phone calls.

No need for separate phone wiring – use computer network:

A VOIP phone system allows you to connect hardware phones directly to a standard computer network port (which it can share with the adjacent computer). Software phones can be installed directly onto the PC. This means that you do not need to install & maintain a separate wiring network for the phone system, giving you much greater flexibility to add users/extensions. If you are moving into an office and have not yet installed phone wiring, you can save significantly by just installing a computer network.

No vendor lock-in:

VOIP phone systems are open standard – all modern IP PBX systems use SIP as a protocol. This means that you can use almost any SIP VOIP phone or VOIP gateway hardware. In contrast, a proprietary phone system often requires proprietary phones, designed specifically for that phone system and proprietary expansion modules to add features and lines.

Scalable:

Proprietary systems are easy to outgrow: Adding more phone lines or extensions often requires expensive hardware upgrades. In some cases you need an entirely new phone system. Not so with a VOIP phone system: a standard computer can easily handle a large number of phone lines and extensions – just add more phones to your network to expand!

Better customer service & productivity:

Because calls are computer based, it is much easier for developers to integrate with business applications. For example: an incoming call can automatically bring up the customer record of the caller, dramatically improving customer service and cutting cost. Outbound calls can be placed directly from Outlook, removing the need for the user to type in the phone number.

Software based Phones are easier to use:

It is often difficult to use advanced phone system features such as conferencing on proprietary phones. Not so with software based SIP phones – all features are easily performed from a user friendly windows GUI.

More features included as standard:

Because a VOIP phone system is software based, it is easier for developers to improve feature sets and performance. Therefore most VOIP phone systems come with a rich feature set, including auto attendant, voice mail, call queuing and more. These options are often very expensive in proprietary systems.

Better control via better reporting:

VOIP settings store inbound and outbound call information in a database on your server, allowing for much more powerful reporting of call costs and call traffic.

Better overview of current system status and calls:

Proprietary systems often require expensive 'system' phones to get an idea what is going on in your phone system. Even then, status information is cryptic at best. With VOIP systems you can define which users can see phone system status graphically via a web browser.

Allow users to hot plug their phone anywhere in the office:

Users simply take their phone, plug it into the nearest Ethernet port and they keep their existing number!

Allows easy roaming of users:

Calls can be diverted anywhere in the world because of the SIP protocol characteristics

For more information about the benefits of an IP PBX, visit the IP PBX FAQ at <http://www.3cx.com/PBX/IP-PBX-faq.html>

How an IP Phone system works

A VOIP Phone System, also referred to as an IP PBX, consists of one or more SIP standard based phones, an IP PBX server and optionally a VOIP Gateway. The IP PBX server is similar to a proxy server: SIP clients, being either soft phones or hardware based phones, register with the IP PBX server, and when they wish to make a call they ask the IP PBX to establish the connection. The IP PBX has a directory of all phones/users and their corresponding sip address and thus is able to connect an internal call or route an external call via either a VOIP gateway or a VOIP service provider.

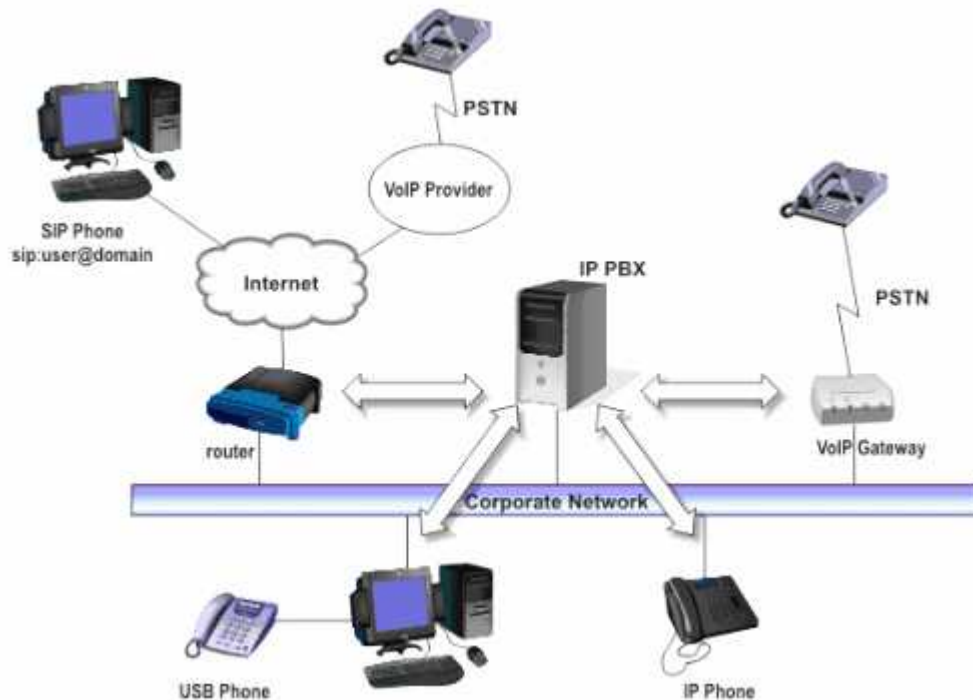


Figure 1 - VOIP Phone System Overview

The image illustrates how an IP PBX integrates on the network and how it uses the PSTN or Internet to connect calls.

For more information about how an IP PBX works, visit the IP PBX FAQ at <http://www.3cx.com/PBX/IP-PBX-faq.html>

SIP phones

A VOIP phone system requires the use of SIP phones. These phones are based on the Session Initiation Protocol (SIP), an industry standard to which all modern IP PBXs adhere. The SIP protocol defines how calls should be established and is specified in RFC 3261. Because of SIP, it is possible to mix and match IP PBX software, phones and gateways. This protects your investment in the phone hardware. SIP phones are available in several versions/types:

Software based SIP phones

A software based SIP phone is a program which makes use of your computers microphone and speakers, or an attached headset to allow you to make or receive calls. Examples of SIP phones are the included 3CXPhone or X-Lite from Counterpath.

Hardware based SIP Phones



Figure 2 - A hardware sip phone

A hardware based SIP phone looks like and behaves like a normal phone. It's actually a mini computer that connects directly to the computer network. Because they have an integrated mini hub, they can share a network connection point with a computer, eliminating the need for an additional network point for the phone. Examples of hardware SIP phones are GrandStream GXP-2000 or SNOM 320.

Analogue phones / Fax machines using an ATA adapter



Figure 3 - An ATA adapter

If you want to use your current phone or analog fax machine with the VOIP phone system, you can use an ATA adapter. An ATA adapter allows you to plug in the Ethernet network jack into the adapter and then plug the phone into the adapter. This way your old 'regular' phone will appear to the VOIP phone system software as a SIP phone.

3CX Phone System editions

3CX Phone System is available in 4 different versions – a Free Edition, a MINI edition, a Small Business Edition, a Pro Edition and an Enterprise edition. A detailed feature

comparison between versions is available at this location: <http://www.3cx.com/phone-system/enterprise-features.html>

Pricing information and ordering information can be found here: <http://www.3cx.com/ordering/index.php>

What's new in 3CX Phone System V8?

All versions

- New Phones management page lists all connected IP phones and allows you to re provision or reboot more than one or all phones remotely.
- Phones management page allows you to easily launch a phone's admin interface
- Ability to edit provisioning files for phones
- Ability to provision FXS gateways
- Ability to forward a voice mail to another user
- Ability to trigger a call back to the person who left a voice mail message
- Setup wizard now allows easy setup of VoIP provider
- Added support for Beronet Gateway cards
- Secure RTP – ability to encrypt voice if phone supports it.
- Improved performance of system.

Small Business, Pro and Enterprise editions

- 3CX Assistant includes integrated text chat – message other extension users
- Improved MyPhone interface with Missed Calls, Outbound and Inbound calls
- Ability to black list callers
- New easy to use Call Forwarding Interface
- Company and Personal phonebook
- Faster click 2 dial with direct IP phone and 3CXPhone integration
- Ability to run the 3CX Assistant from a remote location via the tunnel
- Ability to trigger recording a call from 3CX Assistant
- Call Barge - in
- Ability to accept or reject calls that are forwarded to an external number. Calls that are rejected or not answered will be sent to the phone system voice mail, not the mobile voice mail
- Two additional Queue strategies: Ring All and Prioritized hunt
- Full CRM integration with Microsoft Outlook & Salesforce.com

- New 3CXPhone integration with 3CX Assistant
- Improved paging performance & multi cast support
- Improved call reporter

3CX Training

3CX maintains an online training course which trains users on all important functions of the PBX via a series of video tutorials. The online training course is free of charge and can be completed in approximately one day.

The training course can be found here: <http://training.3cx.com>

It is also possible for administrators to obtain certification. Certification is free of charge and can be taken here <http://www.3cxacademy.com>

3CX Wiki

Additional documentation about 3CX Phone System can be found at our wiki, located at:
<http://wiki.3cx.com>

2. Installing 3CX Phone System for Windows

System requirements

3CX Phone System for Windows requires the following:

- Windows XP Pro, Vista Business (Home editions are not supported), 2003 server or 2008
- Port 5060 (SIP), 5090 (Tunnel - Optional) to be opened
- Port 5480, 5482, 5483, 5485, 5486, 5487 to be free
- One port for IIS, 3 if using Cassini (5000, 5481, 5484)
- .NET Framework version 2.0 or higher
- 1 Gigabyte Memory or higher, Pentium 4 processor or up
- Internet explorer v7 & up, Firefox 2 & up

Recommended

- 3CX supported SIP Phones (<http://www.3cx.com/sip-phones/index.html>)
- 3CX supported VOIP Gateways (<http://www.3cx.com/voip-gateways/>)
- 3CX supported VoIP Providers
- If you are using a VOIP provider, you must have a firewall that is configured to do static port mapping. A static IP is highly recommended also! For more information see: <http://www.3cx.com/support/nat-firewalls.html>
- You will need to have a good basic understanding of Windows Networking.

Run set-up

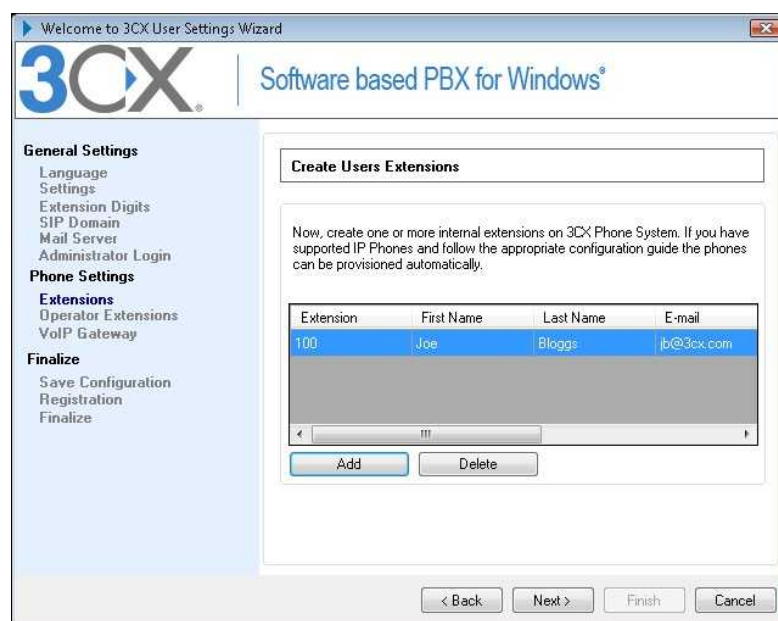
1. Download the latest version of 3CX Phone System from <http://www.3cx.com/ip-pbx/downloadlinks.html>. Run set-up by double-clicking on the set-up file. Click 'Next' to start installation.
2. You will be asked to review and approve the license agreement, as well as to choose an installation location. 3CX Phone System will need a minimum of approximately 50 MB of free hard disk space. You will need to reserve additional space to store voice mail files & prompts.
3. Select the installation location and click Next
4. You will be asked whether you wish to use IIS or the in-built web server, Cassini. Cassini is more light weight, and independent of Windows updates or other IIS applications. Its only draw back is that when using the management console display is a bit slower, although overall phone system performance is NOT affected. Note that

on Windows XP, IIS has a default 10 connection limit, although this can be increased to 40 using a script (<http://www.codinghorror.com/blog/archives/000329.html>)

5. Click 'Install' to start the installation of 3CX Phone System. Setup will now copy all files and install the necessary Windows services. After set-up has completed copying files and installing the services, set-up will run the 3CX Phone System Configuration Wizard to complete the install. Click 'Finish'.

Run the 3CX Phone System configuration wizard

1. The 3CX Phone System configuration wizard walks you through a number of essential tasks that you need to do in order to get your system up and running. After it starts up, it will ask in which language you want the wizard to run.
2. The wizard will ask you how many digits you wish your extension numbers to be.
3. Set-up will prompt you for the FQDN of 3CX Phone System. IP phones contact the phone system server using an IP OR an FQDN. If using an FQDN, then you must specify the FQDN of the server. This value is not relevant if you specify the IP of the server in the phone configuration. For more information, see <http://www.3cx.com/support/sip-domain.html>
4. It will ask you for your mail server name and reply to address. These settings are used to send email notifications, voice mail and faxes. It will ask you for a preferred username and password to be used to logon to the 3CX Phone System management console and manage the phone system.



Screenshot 1 - The 3CX Configuration Wizard

5. Set-up will allow you to create a number of extensions. You can create more later, however you need to create at least one extension. After you have created the extensions, you need to configure your phone sets. 3CX has configuration guides for many popular phones at <http://www.3cx.com/sip-phones/index.html>

6. In the next step it will ask you to specify the operator extension. The operator extension is the default extension used to route inbound calls to. Specify which extension will act as the operator extension. The operator extension will be the default destination for all inbound lines and DID's
7. Now specify the voice mail number – this is the number that users will call to retrieve their voice mail.
8. If you want to use a VoIP provider, you can select the VoIP provider to use, specify server name, proxy (if applicable), account details and rule prefix.
9. 3CX Phone System Wizard will prompt you whether you would like to receive a quote for a complete phone system. If yes, fill in your details and 3CX or a 3CX partner will send you a detailed quote. You can skip this step if you wish.

After the Wizard has completed, you can connect to the 3CX Phone System Management Console by clicking on the management console short cut in the 3CX Phone System program group.

To connect to the management console from a remote machine, start a web browser and enter the name of the machine on which 3CX Phone System is installed, followed by 'Management'. (For example: <http://phone-system/Management>) If you are using Cassini, you have to append the port number, 5481. For example <http://phone-system:5481>

Upgrading from a previous version of 3CX Phone System

It is possible to upgrade a 3CX Phone System 7.1 installation to version 8 as follows:

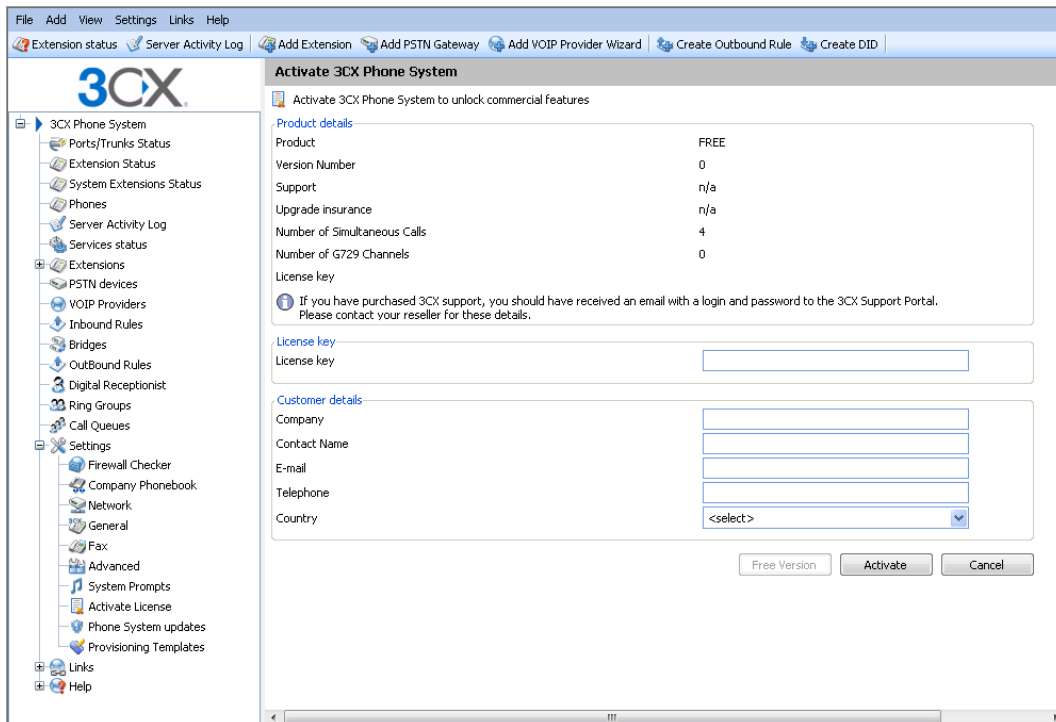
1. Before uninstalling, backup your current configuration using the backup and restore tool located in the 3CX program group.
2. Now uninstall using Add/Remove programs.
3. Install 3CX Phone System v8 – You will be prompted for the backup file by the configuration wizard. This will restore your configuration.

Notes in regards to upgrades from older versions:

- Upgrades of version 3.1, 5, 6.0, 6.1, 7.0 configurations are not supported.
- In the case of an upgrade from 7.0 or 6.1, make a backup of your old configuration. Install v7.1 on to a separate machine, restore the backup and then perform another backup. Then restore that 7.1 backup to v8.
- If you upgraded and switched the type of web server that you used, you must go into 'Phones' node, select all phones and select the 'Re-provision' button. Also you will need to change the URL in each phone. You can either do this manually for each phone according to the configuration guides, or automatically using 'option 66' in your DHCP server. See the chapter on provisioning for more information.

Activating 3CX Phone System

If you have purchased a MINI, Small Business, Pro or Enterprise version then you can activate your license by going to the General > Activate License page in the 3CX Management Console.



Screenshot 2 - Activating your license

Enter your license key, Company, Contact Name, E-mail, Telephone and Country and click on "Activate" to activate your license. This information will be sent to our license key server and your license key and installation will be activated. You will need to do this each time you re-install 3CX Phone System or install an upgrade.

Firewall configuration

There are 3 scenarios in which you will need to update your firewall configuration:

1. If your PBX server is behind a NAT / firewall and you intend to use a VOIP provider
2. If you intend to use remote extensions.
3. If you are using a firewall on the PBX server itself, for example Windows Firewall

3CX Phone System behind a NAT / firewall

The best place for 3CX Phone System is on a machine behind the firewall. This configuration is easier and more secure. If you will just use PSTN lines, you don't even need to make any changes to your firewall configuration.

If you intend to use a VOIP Provider, then you will need to open the following ports to allow 3CX Phone System to communicate with the VOIP Provider:

- Port 5060 (UDP) for SIP communications (send & receive) **MUST BE STATICALLY MAPPED**. See sample firewall configuration at <http://www.3cx.com/support/linksys-configuration.html>
- Port 5090 (TCP) for the 3CX Tunnel (if you plan to use remote extensions)
- Port 9000-9003 (or higher) (UDP) (send & receive) for RTP communications, which contain the actual call. Each call requires 2 RTP ports, one to control to call and one for the call data. Therefore, you must open twice as many ports as you wish to support simultaneous calls via the VOIP provider. For example, if you want to allow 4 people to make calls via the VOIP provider simultaneously, you must open port 9000 to 9007.

Note that the above port ranges are the default ports in 3CX Phone System. You can adjust these ports from the Management Console, in the General > General Settings page. In this page, you can configure the ports to be used for internal calls, and the ports to be used for external calls being made via a VoIP provider.

Firewall installed on the 3CX Phone System machine

If your phone system is running on the internal network behind a firewall, it is not necessary to enable the firewall on the machine running 3CX Phone System. However if you wish to do this, then you will have to open the following ports:

- Port 5060, 5480, 5482, 5483, 5485, 5486, 5487 for 3CX Phone System
- Port 5090 (TCP) for the 3CX Tunnel (if you plan to use the tunnel for remote extensions)
- Port 7000 to 7500 (UDP) for internal leg of calls to VoIP providers or external extensions.
- Port 9000-9003 (UDP) for the external leg of calls to VoIP providers or external extensions. Each call requires 2 RTP ports, one to control the call and one for the call data. Therefore, you must open twice as many ports as you wish to support simultaneous calls via the VOIP provider. For example, if you want to allow 4 people to make calls via the VOIP provider simultaneously, you must open port 9000 to 9007.

Note that the above port ranges are the default ports in 3CX Phone System. You can adjust these ports from the Management console.

The 3CX Extension user manual

The 3CX Extension manual explains users how they can use 3CX Phone System – it includes explanations of how to use 3CX Assistant, the voice mail system, call parking, the MyPhone user portal and so on. The latest manual can be downloaded from here:

<http://www.3cx.com/manual/3CXExtensionmanual8.pdf>

The 3CX Assistant

The 3CX Assistant is a compact utility which allows you to easily manage your extension with a few mouse clicks – rather than via a cryptic and limited phone interface. 3CX Assistant is not a phone – it works in tandem with an existing IP hardware phone, soft phone or even an analog phone. For more information about the 3CX assistant and how to install it, please refer to chapter ‘3CX Assistant’. The 3CX assistant can be downloaded here:

<http://www.3cx.com/phone-system/downloadlinks.html>

The 3CX Assistant install includes 3CXPhone, which can optionally be installed.

System prompts

3CX Phone System ships with a US English prompt set by default. Prompts are voice files that are played by the system to callers and users of the system. For example, when a user picks up their voice mail, the system prompts will instruct the user what buttons to press. To change the system prompts to a different language:

1. Go to the Settings > Phone System Updates page and then select the prompt set you wish to use from the ‘System Prompts’ section. The prompt set will be downloaded.
2. Now go to the Settings > System prompts node and click on the ‘Manage Prompt sets’ button at the top of the screen.
3. Select the prompt set you have downloaded and click on ‘Set As Current Promptset’. The system will now use this new prompt set.

3. Creating Extensions

Introduction

After you have installed 3CX Phone System, you will need to follow a number of steps to complete the setup:

1. Create 3CX Phone System extensions.
2. Configure software and/or hardware SIP phones.
3. Configure phone lines.

Start-up the 3CX Management console from the 3CX program group, or point your web browser to the Management Console by entering the name of the machine, for example: <http://phone-system/Management>.

A login page will appear. Enter the Username and Password that you provided during setup, select the language you would like to use and then click on the 'Login' button.

Note: The management console is available as a Windows application or as a web based application. Naturally the Windows based application is faster.

Adding Extensions

To add an extension, click on 'Add Extension' from the toolbar.

Edit Extension-100

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

General | Forwarding Rules | Phone Provisioning | Other | Blacklist

User Information

Specify extension number, name, and email address for voicemail notifications and fax delivery.

Extension Number: 100

First Name: Nick

Last Name: Galea

Email address:

Authentication

The authentication ID and Password are used by the phone to authenticate with 3CX Phone System and match the ID and Password set on the SIP phone. If the phone has a user id field enter the extension number.

ID: 100

Password: ***

Voice Mail Configuration

If you are unable to answer a call, you can allow voice messages to be taken

Enable Voice mail:

Play Caller ID:

Read out date/time of message: Read in AM/PM format

PIN Number (used by 3CX Assistant): ***

Email Options: Send email notification only

OK Cancel Apply

Screenshot 3 - Adding an extension

User and Authentication information

1. Enter the extension number, first and last name and the Email address (Optional) of the user. The email address will be used for voice mail notifications and as the default SIP ID. You can leave the field empty if you wish.

2. Now specify an authentication ID and password:

- ID – The SIP 'Username'. e.g. 200.
- Password –The SIP Password (password can be hidden from the user).

The Authentication ID and password fields set in the software or hardware phone must match the above fields! In addition the extension number must be entered in the phone as well in a field called Account, User ID or something similar to that. The exact field names differ per vendor. For detailed configuration guides for popular phones see <http://www.3cx.com/sip-phones/index.html>

Voice mail options

3. Now configure the Voice mail options.

- Enable voice mail
- Play Caller ID – the voicemail system will play the number of the caller who left the voice message
- Read out date/time of message – the voicemail system will play back the time of the voice message to be played
- PIN number – this pin number is used to protect the voice mailbox and is used by the user to access the mailbox. The PIN number is also used as a password to logon to 3CX Assistant or the MyPhone User portal.
- Email options – You can choose the following e-mail options when a voice mail arrives:
 - No email notification – The system will not send any emails
 - Send email notification only – This option will notify the user that there is a new voice mail. However the e-mail will not contain the voice mail, the voice mail will have to be retrieved via the phone.
 - Send vmail as attachment – This option will send an email and attach the voice mail as a WAV file to the e-mail. It will still leave the voice mail in the voice mail box just in case you wish to pick up the voice mail via the phone (by dialling 999).
 - Send as attachment and delete from mbox – will send an email with the voice mail attached AND delete the voice mail from the voice mailbox on the 3CX

server. This avoids the user having to delete the voice mail from 2 locations, i.e. from the email inbox and from the 3CX voice mailbox.

4. Click OK to create the extension.

Configuring the SIP phones

After you have created the extension, a summary page will appear, which shows the information that the SIP phone will need:

- Proxy server IP or FQDN: **Host name of 3CX Phone System**
- User ID: **Extension number created**
- Authentication ID: **As specified in Authentication ID field**
- Password: **As specified in Authentication password field**

This information can be entered into the phones via their web interface, or can be provisioned to the phone. For the latest configuration guides how to configure an IP phone via their web interface, go to: <http://www.3cx.com/sip-phones/index.html>

For information how to automatically provision your phones with the correct settings, see the next chapter on provisioning for more information.

Importing extensions

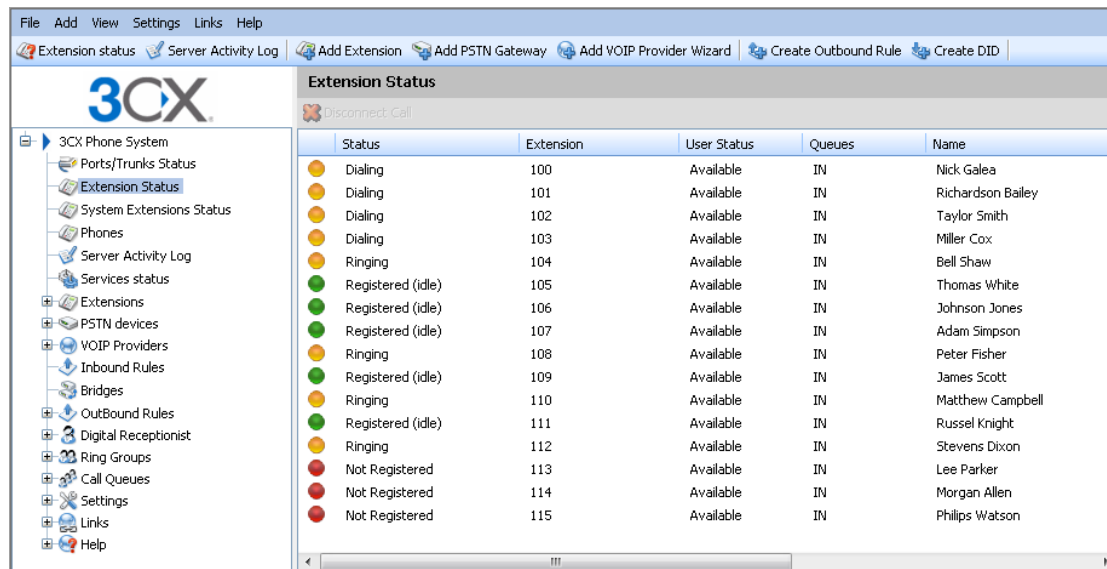
If you need to create a large number of extensions it is handier to bulk import the extensions. To do this, create a spreadsheet with columns for each field that you wish to import. Please find a detailed description of fields as well as a link to a sample import file at

<http://wiki.3cx.com/documentation/general/import-extension-list>

Testing your setup

As soon as you have created at least 2 extensions, and configured the IP phones to work with those extensions, you can test whether your setup is working correctly. To do this:

1. Check Phone System status with the status monitor



Status	Extension	User Status	Queues	Name
Dialing	100	Available	IN	Nick Galea
Dialing	101	Available	IN	Richardson Bailey
Dialing	102	Available	IN	Taylor Smith
Dialing	103	Available	IN	Miller Cox
Ringing	104	Available	IN	Bell Shaw
Registered (idle)	105	Available	IN	Thomas White
Registered (idle)	106	Available	IN	Johnson Jones
Registered (idle)	107	Available	IN	Adam Simpson
Ringing	108	Available	IN	Peter Fisher
Registered (idle)	109	Available	IN	James Scott
Ringing	110	Available	IN	Matthew Campbell
Registered (idle)	111	Available	IN	Russel Knight
Ringing	112	Available	IN	Stevens Dixon
Not Registered	113	Available	IN	Lee Parker
Not Registered	114	Available	IN	Morgan Allen
Not Registered	115	Available	IN	Philips Watson

Screenshot 4 - The Status monitor

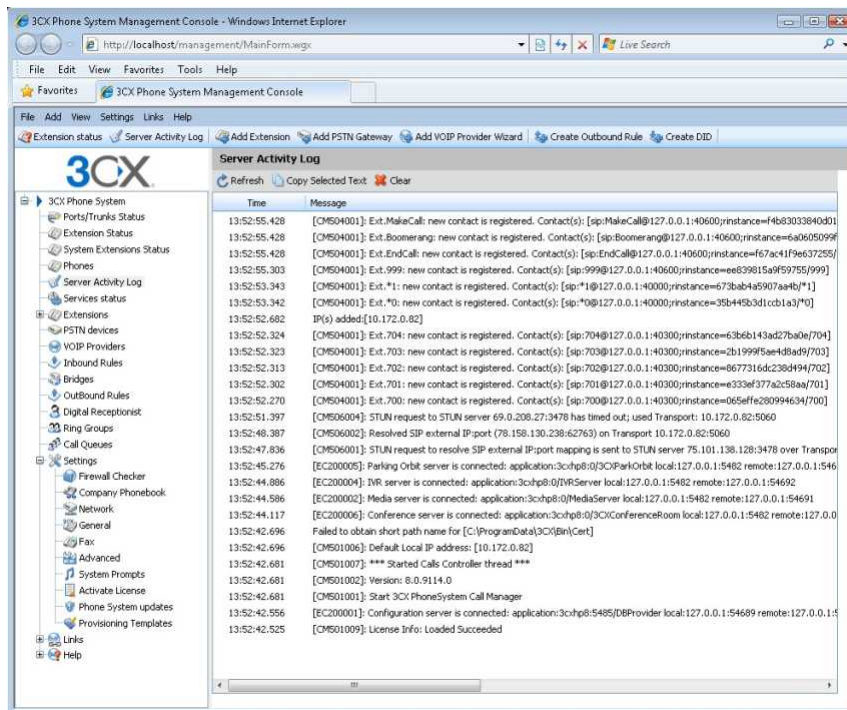
Load up the 3CX Phone System Management console, and click on the 'Extension Status' node. Check that all extensions are listed and are 'Registered'.

If the phone is listed as 'Not registered' then the extension has been created, but the phone has not registered itself with the system. This could be because the device is off, or because the SIP credentials are incorrect. Check whether you have entered the Extension Number, Authentication ID and Authentication Password in the right fields. Another cause could be that you have a firewall enabled on the machine running 3CX Phone System and that it is blocking communications between the server and the phone.

2. Place a call to another extension

If the status shows that the phones have registered correctly, you can make a call to another extension. The status monitor will show that you picked up one extension and are calling another.

3. Troubleshooting tips



Screenshot 5 - 3CX Phone System Activity Log

If dialling another extension does not work, click on the **Server Activity Log** node. This screen shows the activity log of the server, and the log messages can indicate the reason of an error condition. For more information about the server status log, visit <http://wiki.3cx.com>.

4. Phone Provisioning

How provisioning works

Phone provisioning refers to the process of automatically configuring IP phones on a network by having them retrieve the phone settings from the 3CX Phone System server. This method is preferable to manual configuration because it allows you to manage the phone system and phones from a central place, namely the 3CX Management Console.

When using phone provisioning, it is easy to change extension passwords, BLF lights and so on because you can do it from the management console and then reboot the phone and push out the changes without having to look at the configuration of the phone itself.

Provisioning your phones is a two step process:

1. Create the extensions and specify the phone model and its MAC address
2. Configure the phones to retrieve the configuration from the provisioning directory on the 3CX Phone System server.

3CX will prepare a configuration file, usually identical to the phone's MAC address, and publish it via HTTP. The phone will be configured to retrieve its configuration from the 3CX Phone system HTTP server and automatically configure itself.

Step 1 – Configuring the extensions for provisioning

Edit Extension-100

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

General Forwarding Rules Phone Provisioning Other Blacklist

Provisioning

Provisioning ensures the phone settings are centrally retrieved, this limits the amount of time spent and information needed to be configured on each phone.

MAC Address	<input type="text" value="000413236897"/>	?
Model	<input type="text" value="Snom 360"/>	?
Select Interface	<input type="text" value="10.172.0.87"/>	?

BLF (Busy Lamp Fields)

You can map one or more BLF (Busy Lamp Fields) of your phone to particular extensions in order for the user of this phone to see the status of those extensions on his phone. ?

1	<input type="text" value="100 Nick Galea"/>	9	<input type="text"/>
2	<input type="text" value="101 Richardson Bailey"/>	10	<input type="text"/>
3	<input type="text" value="102 Taylor Smith"/>	11	<input type="text"/>
4	<input type="text" value="103 Miller Cox"/>	12	<input type="text"/>
5	<input type="text" value="104 Bell Shaw"/>	13	<input type="text"/>
6	<input type="text" value="105 Thomas White"/>	14	<input type="text"/>
7	<input type="text" value="106 Johnson Jones"/>	15	<input type="text"/>
8	<input type="text" value="107 Adam Simpson"/>	16	<input type="text"/>

Screenshot 6 – Provisioning a Phone

After you have created an extension, go to the 'Phone Provisioning' tab of the extension:

1. Enter the MAC address of the phone (which can be found at the bottom of the phone) in the MAC address field.
2. Select the appropriate phone model
3. Select the Network interface (in case your phone system server has multiple network interfaces)
4. Optionally specify any BLF fields that must be automatically provisioned too. Match a BLF button with an extension, so that this button will show the status of that extension. The number of available BLF buttons varies per phone.
5. Click OK to save. The provisioning files will now be created in the provisioning directory. Each time you make a change to the extension, these files will be re-created. For changes to take effect you must reboot the phone (You can do this remotely from the Phones node).

Step 2 – Configure the phone to retrieve the configuration file

Now you need to instruct the phone that it needs to download its configuration from the provisioning directory on the 3CX Phone System server. This can be done in two ways:

1. By configuring it the phone via the web interface– this is a one time operation and makes sense in smaller networks.
2. By using option '66' in your DHCP server so that the phone will obtain the directory when it receives its IP from the DHCP server

Specifying the provisioning URL via the phones web interface

For smaller networks it makes sense to configure the phone to retrieve the provisioning URL via the phones web interface. This only takes a minute for each phone and it is a one time process. The exact procedure is dependent on the model of phone. The links below describe the process for most popular phones. The latest list can be found at <http://www.3cx.com/sip-phones/index.html>:

Aastra: <http://www.3cx.com/sip-phones/aastra-provisioning.html>

Grandstream: <http://www.3cx.com/sip-phones/Grandstream-provisioning.html>

Linksys: <http://www.3cx.com/sip-phones/linksys-spa-9xx-provision.html>

Snom: <http://www.3cx.com/sip-phones/Snom-provisioning.html>

Polycom: phones must be provisioned using option '66' as the phones web interface does not have a way to configure the provisioning directory

Specifying the provisioning URL via DHCP option '66'

For larger networks it is recommended that you specify the provisioning URL to the phones via DHCP option 66. Essentially the phone will then be told where to get its configuration file at the same time it receives its IP and networking info. This makes it

easy to change the provisioning directory later, for example in case you wish to move your phone system to another server.

To use this option, you must configure your DHCP server to provide this information. On this link you will find a guide how to configure DHCP option '66' for Microsoft DHCP servers.

Microsoft DHCP Server: <http://www.3cx.com/sip-phones/DHCP-option-66.html>

If you do not use Microsoft DHCP server, you will need to refer to the documentation of your DHCP server for more information on how to configure this.

Note: Linksys phones can not be provisioned via option 66 and HTTP. A TFTP server would be required. To avoid this, for EVERY phone, in a Browser window, run this URL:

"http://10.0.0.103/admin/resync?http://10.0.0.11/management/provisioning/spa\$PSN.cfg"

Where 10.0.0.103 is the IP address of the Linksys phone, and 10.0.0.11 is the IP of 3CX Phone System. You only need to do once.

Provisioning URL

The location of the provisioning directory, and thus the URL, differs depending on whether you are using Cassini or IIS as the web server for 3CX:

If using IIS

"http://<ip of 3CX>/management/provisioning" – e.g.

http://10.172.0.2/management/provisioning".

If using Cassini,

"http://<ip of 3CX>/provisioning" – e.g. "http://10.172.0.2/provisioning".

Provisioning URL format for each phone

Unfortunately each phone requires the URL to be specified in a certain manner:

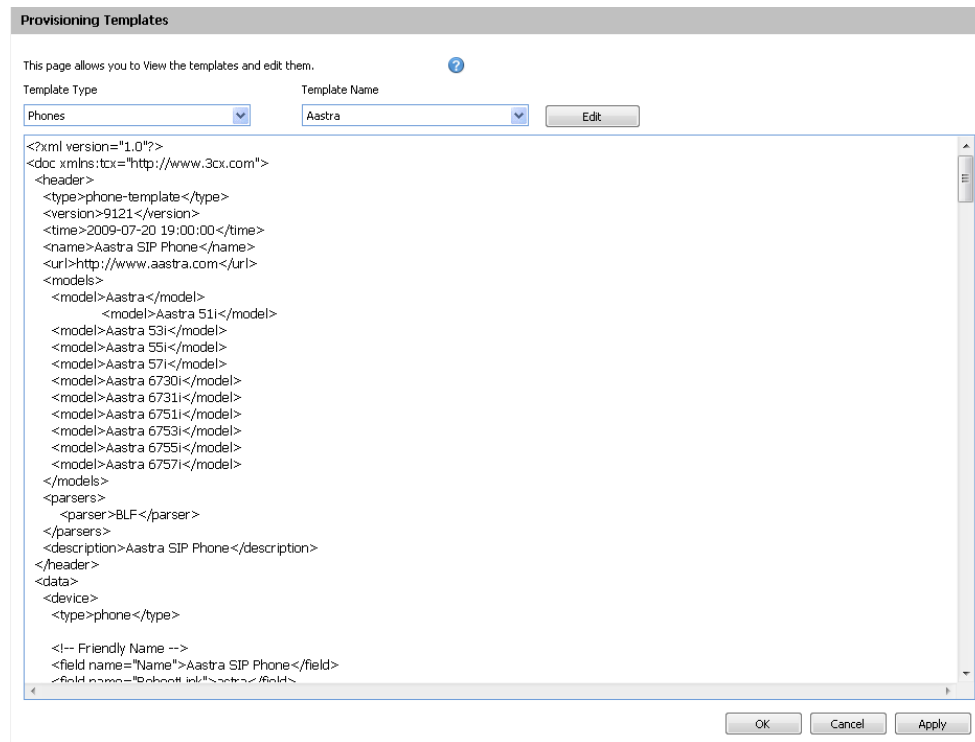
Aastra Phones: "http://10.0.0.11/management/provisioning/"

Grandstream Phones: "10.0.0.11/management/provisioning"

Snom Phones: "http://10.0.0.11/management/provisioning/cfg{mac}"

Polycom Phones: "http://10.0.0.11/management/provisioning/"

Phone provisioning templates



Screenshot 7 - Editing the Phone Provisioning template

The phones are provisioned based on XML templates, and if you are familiar with the model of IP phone that you have, it is possible to edit these templates and add additional configurations that you might need (for example to switch off backlight on the display). These edits will then be added to the provisioning files. To edit the phone provisioning templates:

1. Go to the Settings > Provisioning templates node
2. In Template Type select 'Phones'
3. In 'Template Name' select phone model
4. Edit the template as necessary and click OK to save the changes. To make the changes active, you need to re-provision the phones from the phones node.

Re-provisioning the phones

If you need to re-provision the phones, for example after you have made configuration changes, you can easily do this from the IP Phones node:

1. Go to the 3CX Phone System > Phones node
2. Select the phones that you wish to re-provision
3. Click 'Re-provision phones' - a new provision file will be created
4. Select the phones again and select 'Reboot' to make the new provisioning information active on the phone.

Monitoring your IP phones

Phone Model	Name	User ID	User Password	PIN	IP of Phone	MAC Address
UNKNOWN	NEW	NEW	NEW	NEW	10.172.0.118	000413246674
UNKNOWN	NEW	NEW	NEW	NEW	10.172.0.128	000413246C56
UNKNOWN	NEW	NEW	NEW	NEW	10.172.0.142	000413236897
GrandStream GXP-2000	Grandstream GXP 2000	100	***	***	10.172.0.100:5060	000B820A5E5C
GrandStream GXP-2020	Grandstream GXP2020	101	***	***	10.172.0.101:5060	000B820F3328
Aastra 51i	Aastra 51i	102	***	***	10.172.0.102:5060	00085D1A77D0
X-Lite release 1103k stam	a	103	***	***	10.172.0.133:41674	UNPROVISIONED
Snom	Snom 320	104	***	***	10.172.0.104:2048	000413246679
Linksys SPA-901	Linksys SPA942	105	***	***	10.172.0.105:5060	000E080F8D10
Polycom SPIP320	Polycom 320	106	***	***	10.172.0.106:5060	0004f214b038
Polycom SPIP550	polycomip660 Polycom	107	***	***	10.172.0.107:5060	0004f215ec89
Polycom SPIP430	Polycom Soundstation	109	***	***	10.130.0.104:5060	0004f221ad1a
Linksys SPA-921	Linksys SPA921	110	***	***	10.172.0.110:5060	000E080B5986
Polycom SPIP650	Polycom2 Soundstation	111	***	***	10.130.0.106:5060	0004f2229984
Aastra 55i	Aastra 55i	112	***	***	10.172.0.112:5060	00085D1B6048
T28 2.3.0.10	Yielink AY	114	***	***	10.172.0.122:5062	UNPROVISIONED
Cisco-CP7912/8.0.1-0604	dummy	115	***	***	10.172.0.125:5060	UNPROVISIONED
Snom	MAIN2 360	116	***	***	10.172.0.116:2048	00041323689a
snom320/7.3.14	Snom production	117	***	***	192.168.1.24:2048	UNPROVISIONED
Aastra SIP-DECT (SW-Ver	AastraDect	118	***	***	10.172.0.202:5060	UNPROVISIONED
Linksys SPA-962 + SPA-9	Linksys SPA962	119	***	***	10.172.0.119:5060	000E080D5526
3CXVoipPhone 4.0.8913.0	SuperAdministrator	120	***	***	127.0.0.1:51326	UNPROVISIONED
Cisco-CP7940G/8.0	SuperAdministrator	120	***	***	10.172.0.120:5060	UNPROVISIONED

Screenshot 8 - The phones node

3CX Phone System provides an easy way to monitor and manage your phones network wide. The 'Phones' node in the 3CX Management console allows you to:

- View all phones available in the network
- Quickly view IP and Mac address of each phone
- Remote reboot one or all of the phones
- Re-provision the phones (after you have made a change you can update the provisioning files and reboot the phones to have the changes take effect)
- Launch the admin interface of the phone
- Monitor security of extension password and PIN. Weak extension passwords and PIN's are the most common cause of security breaches

5. Adding PSTN line(s) via a VOIP Gateway

Introduction

External calls are received and made on PSTN phone lines. A traditional PBX requires you to connect the PSTN lines to the PBX hardware box; however in the case of 3CX Phone System you have more options:

- Connect PSTN lines (physical phone lines) to a VoIP Gateway situated on your internal network.
- Connect PSTN lines to a VoIP add in card, such as a Sangoma or Beronet, installed on the 3CX Phone System machine or on another machine.
- Use a 'hosted' phone line from a VOIP Service Provider. In this case the VOIP service provider has connected external lines to a VOIP Gateway and allows you to access these lines via your internet connection.

To make & receive external phone calls via your regular phone lines, you will have to buy and configure a VoIP gateway or VoIP add in card. This chapter explains what they are and how to configure them.

What is a VoIP Gateway or VoIP add in card?

A VoIP gateway is a device which converts telephony traffic into data, so that it can be transmitted over a computer network. In this manner PSTN/telephone lines are "converted" to SIP extensions, allowing you to receive & place calls via the regular telephony network. VOIP Gateways exist for analog lines as well as BRI, PRI/E1 lines and T1 lines. VoIP cards do the same thing, but are add in cards that are installed into an existing computer.

There are lots of VoIP gateways and VoIP cards available today at competitive prices. Analog VOIP gateways start at as little as \$100. Be sure to use a supported device - a list of supported gateways & cards can be found on <http://www.3cx.com/voip-gateways/index.html>.

PSTN line configuration overview

Just as it is necessary to configure a phone to register with the phone system, it is necessary to configure the VoIP gateway or card to register its lines with 3CX Phone System. Each port gets a SIP user ID, Password and virtual extension number. To the IP PBX, the PSTN line appears just like any other SIP extension.

In the next sections we will now describe step by step how to configure a VOIP Gateway for use with 3CX Phone System.

Definitions

In this manual and in the 3CX Phone System Management Console a number of terms are used:

Port – a port is a physical port on a gateway or VoIP card. In the case of an analog line, one port equals one voice channel. In the case of BRI ISDN, one port is 2 voice channels, and in the case of E1 or T1 ports, each port represents 30 and 23 channels respectively.

Virtual Extension Number – this is an extension number assigned to a port. This virtual extension number is used by the port, together with the authentication ID and the authentication password, to register with 3CX Phone System.

Port identification – this is a field used to identify the port. You can enter the PSTN line number here, or simply a text string such as ‘sales’

Step 1: Create the PSTN gateways in 3CX Phone System

The first step is to create the PSTN gateway in the 3CX management console.

The screenshot shows the 'PSTN devices' section with the 'Add Gateway Wizard' active. The 'Add PSTN Gateway' form is displayed with the following fields and values:

Name	Analog Gateway
Brand	Patton
Model	SN-4114 4-port FXO (Firmware R5.x)
Description	4-Port Analog FXO
URL	http://www.patton.com

Below the form, there is a note: "More vendor supported gateways can be found here: <http://wiki.3cx.com/gateway-configuration/vendor-supported>". At the bottom of the form are 'Cancel' and 'Next >' buttons.

Screenshot 9 – Choosing gateway template

1. In the 3CX Phone Management console menu click on Add > PSTN Gateway.
2. In the name field, enter a friendly name for the VOIP gateway. Now choose the gateway brand and model that you are using from the list. If it is not listed, select generic. Note that we will not be able to support you in using this gateway. Now click “Next”.
3. Depending on the gateway you selected, you might be asked additional options, such as what country the device will be connected in and so on.

The screenshot shows the 'PSTN devices' section with the 'Specify VoIP Gateway Details' wizard active. The 'VOIP Gateway' form is displayed with the following fields and values:

Gateway Hostname or IP	
Gateway Port (default is 5060)	5060
Number of ports	4
Type	Analog

At the bottom of the form are '< Back' and 'Next >' buttons.

Screenshot 10 - Specifying VoIP Gateway details

- Now enter the host name or IP of the VOIP Gateway in the 'Gateway Hostname or IP' field, and specify the SIP Port on which the gateway is operating. By default this is 5060.
- Specify the number and type of ports the gateway supports, i.e. analog, BRI, PRI or T1. This will set up one account for each port and enable the corresponding number of calls/lines for that account. An analog line supports 1 call; a BRI port supports 2 calls, an E1 (PRI) 30, and a T1 (PRI) 23. For example, if you specify 1 * T1 port, it will create one SIP account which can handle up to 23 calls. If you wish to have each line individually addressed, simply select 23 * Analog lines. The other options, such as Registration Settings, Other options etc. have been preconfigured for your gateway model and do not need to be changed. Click Next to go to the next screen.

PSTN devices

Create ports

The following ports will be created in the "Create Ports" screen. You can edit the Port identification and authentication settings before they are created. Note that the Port identification is used for identification purposes, and the internal line number is used by 3CX Phone System to address the line connected to the port on the VOIP Gateway. Therefore the Internal Line Number range should be different from the extension number range. You can configure to which extension incoming calls should be routed based on whether they are inside or outside office hours (inbound route).

Remove selected	Virtual extension	Authentication ID	Authentication Password	Channels	Port Identification	Inbound Route Day	Inbound Route Night
<input type="checkbox"/>	70002	70002	70002	1	70002	101	101
<input type="checkbox"/>	70003	70003	70003	1	70003	101	101
<input type="checkbox"/>	70004	70004	70004	1	70004	101	101
<input type="checkbox"/>	70005	70005	70005	1	70005	101	101

< Back

Next >

Screenshot 9 - Adding PSTN Lines

- Click Next. The individual ports will be 'created' and displayed in a columnar format.

Virtual extension number – In effect the VOIP Gateway "converts" each line/port to an extension, so that the phone system can receive and forward calls to it. The virtual extension number is a number assigned to it by 3CX Phone System so that it can address it as an extension. There is no need to change this field.

Authentication ID & Password: These values are used to authenticate the ports with 3CX phone system. **These values must match the settings configured in your gateway!** By default the ID and password have been set to the Virtual extension number.

Channels: The Channels field allows you to specify how many simultaneous calls this port supports. An analog line supports 1 call; a BRI port supports 2 calls, an E1 (PRI) 30, and a T1 (PRI) 23. If you prefer to address each line individually, you can create additional SIP accounts and change the number of channels supported by each account to 1. Note that your VOIP Gateway must support this – especially higher density VOIP Gateways are easier to configure if you use one account for all lines connected to a port.

Port Identification – This field specifies the actual PSTN phone line number and is used for identification purposes only.

Inbound Route: If the port will receive inbound calls, you can specify to which extension, ring group or digital receptionist a call must be routed.

- On the next page, you will be asked for a prefix so as to create an outbound rule for this device. Enter the dialling prefix in the "Calls to numbers starting with (prefix)" text box. To make calls via this gateway, precede the number to be dialled with this prefix. Click

Finish to create the lines. You can change line options of lines that have been created by going to the Lines node.

Step 2: Configure the VoIP gateway

After you have created the PSTN Lines/ports, a summary page will appear, which shows the information that you need to enter into the VOIP Gateway:

- Proxy server IP or FQDN: **Host name of 3CX Phone System**
- User ID: **Virtual Extension number**
- Authentication ID: **As specified in Authentication ID field**
- Password: **As specified in Authentication password field**

The Virtual extension number, authentication ID and authentication password must be duplicated for each port that you have on your gateway.

For a detailed description how to do this for popular gateways, follow the configuration guides listed on last page of the configuration wizard, or go to: <http://www.3cx.com/voip-gateways/index.html>

Generating a configuration file

For some gateways, 3CX Phone System is able to create a configuration file that can be uploaded to the device and makes configuration automatic. In this case, the button 'Generate config file' button will be active. Clicking the button will generate a text based configuration file which can then be uploaded to the device using the web based interface of the device.

Configuring the PSTN Interface

In some cases you might need to check that the PSTN interface is configured correctly for your country. If in doubt, check with the VOIP Gateway supplier. For example if you have bought a device that is configured for the US but you are using it in the UK, you might have issues such as the device not detecting a hang up by a caller. In these cases you will have to configure the Call Progress Tones for the VOIP Gateway. For more information about this consult the VOIP Gateway supplier and manual.

6. Adding lines hosted by a VOIP Provider

Introduction

VOIP providers “host” phone lines – they can assign local numbers in one or more cities or countries and route these to your phone system. In addition, VOIP providers are often able to give better call rates because they have an international network or have negotiated better rates. Therefore, using VOIP providers can reduce call costs. However be aware that each VOIP call requires bandwidth. VOIP is real time, so it does place a demand on your internet connection. As a rule of thumb, each call will consume approximately 30k-120k per second, depending on which codec you use. This chapter describes with which VOIP providers you can set-up an account and how to configure 3CX Phone System accordingly.

Requirements for using a VOIP provider

If you plan to use a VOIP provider, you will have to have a firewall/router/NAT device that supports **STATIC PORT MAPPINGS**. Often routers will perform port address translation, which will cause problems such as one way audio, failing inbound calls and so on. It is also highly recommended that you have a **static external IP**. If your external IP changes at certain intervals, inbound calls will fail. Please see our wiki for more information and sample router configurations.

Step 1: Create an account with a VOIP Provider

To add a VOIP line you need to have an account with a quality VOIP service provider. 3CX Phone System supports most popular SIP based VOIP service providers and we recommend using one that has been tested by 3CX. 3CX includes pre configured templates for these VOIP providers. Simply click on the ‘Add VOIP’ link to see a list of supported providers.

Note that you can use any SIP based provider but we will not be able to support you in the configuration of the VOIP provider, nor can we guarantee that it will work. Therefore, unless for your country no VOIP provider is listed, use a supported VOIP provider.

Step 2: Add the VoIP provider account in 3CX Phone System

After you have created the VOIP provider account, you will need to configure the account in 3CX Phone System. To do this:

1. In the 3CX Phone Management console menu, select Add > VoIP provider.
2. Enter a friendly name for this VOIP provider account and choose the VOIP provider that you are using. If it is not listed, select ‘Generic VOIP provider’, or ‘Generic SIP Trunk’. A generic SIP trunk is a provider that does not use an authentication ID and

password, but rather uses your IP as authentication. If using a generic provider we will not be able to guarantee that it will work with this VOIP provider. Click Next.

















VOIP Providers

[Add VOIP Provider Wizard](#)

[Add VOIP Provider Wizard](#)

Name of Provider

Choose a Provider:

<input checked="" type="radio"/>		888VoipStore	US
<input type="radio"/>		Actio.pl	PL
<input type="radio"/>		Broadvox GoAnywhere	US
<input type="radio"/>		Broadvox SIP Trunk	US
<input type="radio"/>		CallCentric	US
<input type="radio"/>		Generic SIP Trunk	
<input type="radio"/>		Generic VoIP Provider	
<input type="radio"/>		nettel	DK
<input type="radio"/>		Nexvortex	US
<input type="radio"/>		Prionet	NL
<input type="radio"/>		SoTel SIP Trunk	US CA
<input type="radio"/>		Tweakphone	NL
<input type="radio"/>		VolPOn	UK
<input type="radio"/>		Voip-Unlimited	UK
<input type="radio"/>		Weepee	BE
<input type="radio"/>		Wide VOIP	FR, LU

More 3rd party tested providers can be found here: <http://wiki.3cx.com/voip-provider/3rd-party-supported>

Screenshot 11 - Add VOIP Provider account

- The registrar/Proxy Hostname will be pre-filled. Compare these with the details that you have received from your VOIP provider and check that these are indeed correct. Depending on the VOIP provider that you are using, some fields will be disabled. This means you do not need to change them. Click Next to continue.
- Now enter the VOIP provider account details. In the VOIP External number field, enter the VoIP line number that has been assigned to. Then enter the Authentication ID/user name and password of your VOIP provider account. Specify the number of simultaneous calls your provider supports. Click Next to continue. If you are using a SIP trunk, the authentication ID and password will be greyed out, since authentication is done via IP.
- Now specify how calls from this VOIP provider should be routed. You can specify a different route outside office hours.
- On the next page, you will be asked for a prefix so as to create an outbound rule for this VOIP provider. Enter the dialling prefix in the "Calls to numbers starting with (prefix)" text box. To make calls via this provider, precede the number to be dialled with this prefix.

Note: Frequently the internet facing firewall sitting between 3CX Phone System and the VOIP provider is not correctly configured or is not able to correctly route VOIP traffic. To check the firewall configuration, it is important to perform a firewall check using the inbuilt firewall checker. You can start it by going to the Settings > firewall checker node. It will connect with the public 3CX firewall check server (firewallcheck.3cx.com) and will check that it can send and receive UDP data on port 5060 (for SIP), port 3478 (for STUN) and port 9000 to 9016 (for up to 8 simultaneous calls, default setting). If it can, the firewall check will succeed. If the firewall check fails, you will not be able to reliably make and receive VoIP provider calls and you will have to edit your firewall configuration. Note that we do not provide firewall configuration support.

Specifying a STUN server

Although 3CX includes a default STUN server setting, it is recommended that you specify the STUN server suggested by your VoIP provider as your default STUN server. The STUN server suggested by your VoIP provider is probably closer to you and therefore quicker to reach (less hops). The faster the STUN server responds to requests, the quicker the call will be setup. To specify a STUN Server:

1. In the Management Console, go to Settings > Network node. Now click on the 'STUN server' tab.

STUN server options

Specify STUN server that should be used to resolve public IP automatically. If your machine has a public IP AND CORRECT STATIC PORT MAPPING, you can switch off STUN

Default STUN server address:	<input type="text" value="stun.3cx.com"/>	?	Port:	<input type="text" value="3478"/>	?
Default second STUN server address:	<input type="text" value="stun2.3cx.com"/>	?	Port:	<input type="text" value="3478"/>	?
Timeout time for STUN response (ms):	<input type="text" value="3000"/>	?			
Query STUN server every (sec):	<input type="text" value="1200"/>	?			

Screenshot 11 - Specifying STUN server

2. In the edit box 'Primary STUN server' specify the STUN server suggested by your provider, for example `stun.sipgate.net:10000`
3. You can specify an alternate backup server, or leave the `stun2.3cx.com` as the backup STUN server. This address will be used if the primary STUN server is down. Click OK to exit and save the settings.

DID's and VoIP providers

If your VoIP provider has supplied you with DID's AND inbound call identification is based on destination/dialled number, it will be necessary to configure the Source Identification rules for the VoIP Provider. This is because the identification number will change depending on which DID number has been dialled. In this case you have the following options:

1. If possible, consider using a different source identification rule, based on IP or on Authentication ID.

2. Alternatively, enter all the possible DID's associated with your account. If it is a large range, it is best to enter this as a mask using a wild card character (*). This helps improve system performance. For example, if you have a range of 999 DID numbers, e.g. 2456000 to 2456999, you can enter 2456*

Both options can be configured by editing the VoIP provider account and going to the "Source ID" tab. In the source ID tab you configure how inbound calls are identified.

To create a different source identification rule:

1. In the section "Call Source identification", select the appropriate SIP field and the corresponding variable or value.

To specify the DID's associated with this account

1. Enable the option "Source Identification by DID"
2. Now specify the SIP field that contains the DID numbers
3. Specify the DID numbers. The option 'Add Mask' allows you to add a mask that covers a range of DID numbers. Add DID number will bring up a dialog allowing you to select from DID numbers that you have already inputted in the DID tab of the VoIP provider. Click OK when done.

Note: Configuration is necessary because source identification based on DID and routing based on DID are two different things and thus require separate configuration.

7. Creating Outbound Call Rules

Introduction

An outbound rule defines on which gateway/provider an outbound call should be placed, based on who is making the call, the number that is being dialled and the length of the number.

Creating an outbound call rule

Edit Outbound Rule

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

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)

Calls to Numbers with a length of

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.

			Strip Digits	Prepend	
Route	1	<input style="border: 1px solid #ccc;" type="text" value="Voip-Unlimited - UK"/>	<input style="border: 1px solid #ccc;" type="text" value="1"/>		
Route	2	<input style="border: 1px solid #ccc;" type="text"/>	<input style="border: 1px solid #ccc;" type="text" value="1"/>		
Route	3	<input style="border: 1px solid #ccc;" type="text"/>	<input style="border: 1px solid #ccc;" type="text" value="1"/>		

Screenshot 12 - Creating a new outbound rule

To create an outbound rule:

1. On the 3CX Management Console menu, click Add > Outbound Rule, and enter a name for the rule.
2. Now specify for which calls to apply the outbound route. In the 'Apply this rule to these calls' section, specify any of these options:
 - Calls to Numbers starting with – apply this rule to all calls starting with the number you specify. For example, specify 0 to specify that all calls starting with a 0 (usually a prefix) are outbound calls. Callers would dial '0123456' to reach the number '123456'
 - Calls from extension(s) – Select this option to define particular extensions or extension ranges for which this rule applies. Specify one or more extensions separated by commas, or specify a range using a -, for example 100-120

- Calls with a Number length of – Select this option to apply the rule to numbers with a particular digit length, for example 8 digits. This way you can capture calls to local area numbers or national numbers without requiring a prefix.

3. Now specify how the outbound calls should be made. In the 'Make outbound calls on' section, select up to 3 routes for the call. Each defined gateway or provider will be listed as a possible route. If the first route is not available or busy, 3CX Phone System will automatically try the second route.

4. You can transform a number that has been 'caught' by the outbound rule before it is routed to the selected gateway or provider using the 'Strip digits' and 'Prepend' fields:

- Strip digits – allows you to remove 1 or more of the digits. Use this to remove the prefix before it is dialled on the gateway or provider. In the example above, you would specify to remove 1 digit, in order to remove the prefix '0' before it is dialled.
- Prepend – allows you to add one or more digits required by the provider or gateway.

You can set these options per route, since a VOIP gateway connected to the local PSTN might require a different logic than if the call is routed via a VOIP provider.

8. The 3CX Assistant

Introduction

The 3CX Assistant is a compact utility which allows users to easily manage their extension with a few mouse clicks – rather than via a cryptic and limited phone interface.

3CX Assistant is not a phone – it works in tandem with an existing IP hardware phone, soft phone or even an analog phone. Calls are made and answered on your existing phone. 3CX Assistant provides the following functions:

1. **Call Pop-up** - Upon an incoming call, a small dialog will pop up, which allows you to reject a call or transfer a call to another person or to voice mail - with a single mouse click or using drag and drop.
2. **Easy Call Transfer / Park** - When on a call, you can transfer or park a call with a mouse click or via drag and drop, no need to learn dial codes or call transfer procedures on a phone.
3. **Presence** - The status of the other extensions is shown, allowing you to avoid unnecessary calls or call transfers to colleagues.
4. **Click to Call** - Launch calls with a mouse click – double click on an extension to call the person, or enter a name or number in the make call dialog. The call will automatically be launched without requiring you to dial the number. Calls can also be launched directly from your contact management software with the CRM integration module.
5. **Queue monitoring** - View the status of queues that you are a member of. You will see callers waiting in the queue and be able to take a call from the queue.
6. **Text chat** – Message other users using the in-built chat option
7. **Record calls** - you can trigger the recording of a call by hitting the record button
8. **Launch MyPhone** - You can easily launch the MyPhone page from the 3CX Assistant. Authentication will be automatically done using the 3CX Assistant credentials and users will be able to configure call forwarding or view their recent call activity including missed calls.
9. **Phonebook** – 3CX Assistant provides easy access to the company and to the personal phonebook, and allows users to trigger calls simply by typing a name – this will then automatically resolve the number and launch the call.

3CX Assistant will show different information based on whether you are a standard user, a departmental manager or a company manager. In department manager mode you will see calls from anyone in your department. In management mode you will see information

of the entire company. These rights are set by the Phone System administrator in the 3CX Management console.

Installing 3CX Assistant

System requirements

- Windows XP, Vista or higher
- .NET 2.0 framework or higher. Most Windows installations have this already installed. Check by going to Control Panel > Add/Remove Programs. If not, the latest .NET framework can be downloaded from the Microsoft website at <http://msdn.microsoft.com/en-us/netframework/>

Running Setup

1. Download 3CX Assistant here

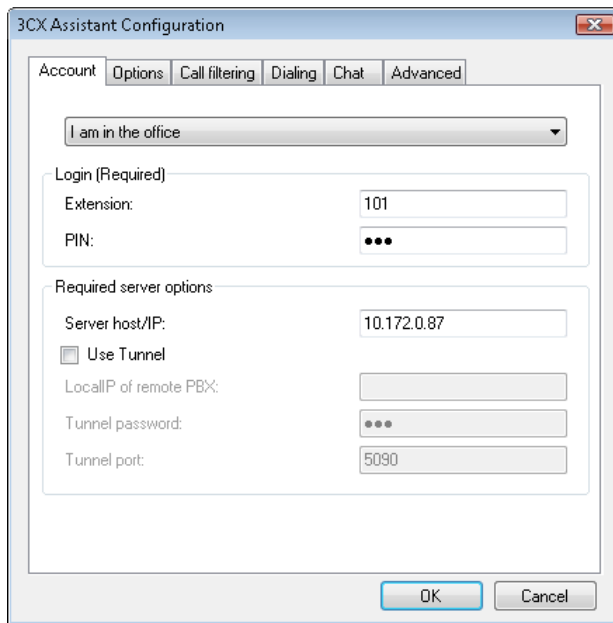
<http://www.3cx.com/downloads/3CXAssistantSetup8.exe>

2. Execute setup and follow the wizard

3. You will be prompted if you wish to install CRM integration. CRM integration allows calls to be launched from Microsoft Outlook and Salesforce.com, as well as pop up the customer record based upon receiving an incoming calls based on the caller ID. CRM integration is documented in a separate manual that you can download from this page: <http://www.3cx.com/phone-system/downloadlinks.html>

4. NOT AVAILABLE IN BETA Setup will prompt you whether you wish to install 3CXPhone. 3CXPhone is a softphone that will allow you to make and receive calls from your computer. If you install 3CXPhone, you can configure 3CX Assistant to integrate with it and launch 3CXPhone automatically for Outbound and Inbound calls.

Configuring 3CX Assistant



Screenshot 12 - 3CX Assistant configuration

The first time you start 3CX Assistant, you must enter your extension details:

1. In the login section, enter your extension number and your voice mail pin number (By default the PIN is the same as your extension number). You can change this from your voice mail menu or from the MyPhone user portal.
2. In the location section, specify In or Out of the office.
3. In the Server options section, specify The local IP or host name of the 3CX Phone System machine.
4. If you are going to use the 3CX Assistant outside of the office, you must also specify:
 - a. The public IP or host name of the 3CX Phone System machine
 - b. Tunnel Password

Tunnel Updating 3CX Assistant network wide

3CX Assistant includes an automatic updating mechanism which saves you having to update the 3CX Assistant installation on each desktop. To make use of this feature:

1. Download 3CX Assistant and check that this is an update you wish to deploy to all your computers.
2. Copy the file to the Application Data directory on the server:

Start/Run

```
%allusersprofile%\3CX\Data\AssistantStore\ClientUpdates_
```

The update will be named tcx.assistant.client.exe

3. 3CX Assistant will check this directory, download the update and then restart.

Note: If the user refuses to perform the update, the user will receive a notification after 30 minutes that there is an update available.

Note: To update the client immediately, select Help, Check for updates.

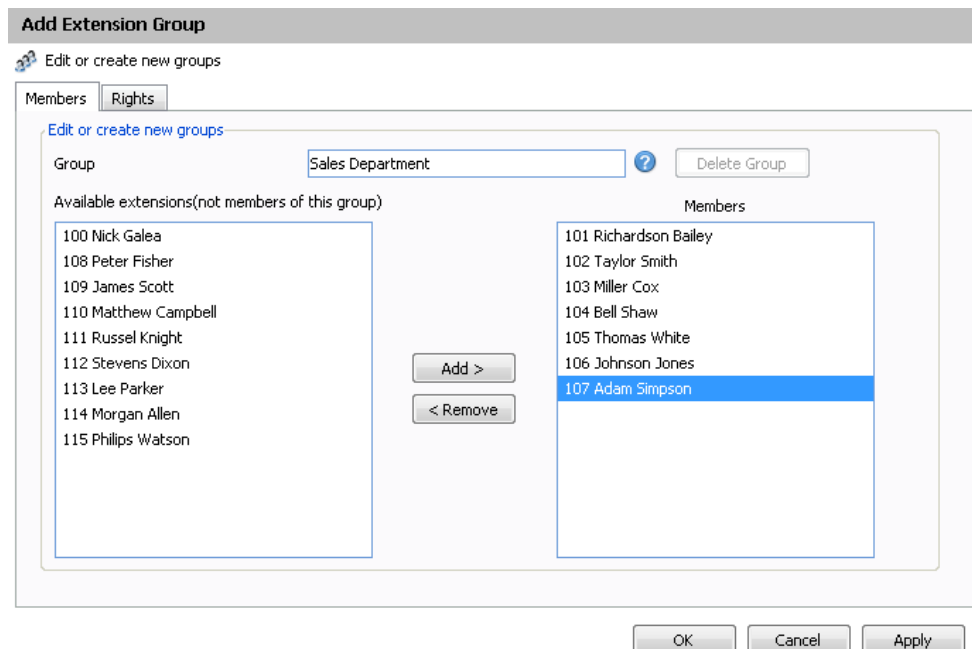
Using 3CX Assistant

A separate user manual exists for 3CX Assistant. This manual can be found here:

<http://www.3cx.com/manual/3CXExtensionmanual8.pdf>

Grouping & Assigning Rights

3CX Assistant can limit what call information is displayed based on extension groups. These extension groups are used to determine what information is shown to whom. In addition they help group the extension for both users and the administrator. To create an extension group:



Screenshot 13 - Adding an Extension Group

1. In the management console, click on the Extensions node and click on the 'Add Extension Group' icon.
2. Now proceed to add Extensions by selecting extensions and clicking on the 'Add' button. Note that Extensions can only be part of one group, so only extensions that have not been assigned to a group yet are listed.

Add Extension Group

Edit or create new groups

Members Rights

Group Administrator

Administrator Extensions: Select Extensions

? Perform operations on calls to users of this group

? Can Barge In on calls to users of this group

User rights

? See call details (call destination, duration) on any active call to group members

? See incoming calls to any group member

? Perform operations (park, transfer, take) on any active call to group members

? Can Barge In on calls to users of this group

OK Cancel Apply

Screenshot 14 - Setting Extension Group Rights

3. Now click on the 'Rights' tab. These settings pertain to what group members can see and do in the 3CX Assistant.
4. You can configure one or more 'Administrators' for this group. Typically this would be a department supervisor. Group administrators will be able to see the call details of everyone WITHIN that group.
5. •Optionally you can enable the Administrator extensions to perform operations on calls of any of the group members. Check the option "Perform operations on calls to users of this group" to enable this.
6. You can also configure what rights regular group members have:
 - See Call Details of other group members
 - See incoming calls to other group members
 - Perform operations on calls of group members.
 - Can Barge in on calls to users of this group
7. Click OK to save group and rights information. Users will need to login and logout of 3CX Assistant to see their new rights reflected.

The Management Extension group

The Management Group is a group that is always present in 3CX Phone System. Extensions assigned to this group can:

1. See the call details of ALL extensions and queues
2. Perform operations on ANY call in the system (Pickup, transfer, Divert, Reject, Barge-in, Park).

To add an extension to the management group:

1. Click on the MANAGEMENT sub-node under the extensions node.
2. Select an extension and click Add. Press Apply/OK to save the changes.
3. Now logout and login with 3CX Assistant to see the additional information.

9. The Digital Receptionist / Auto attendant

Introduction

The digital receptionist feature allows you to answer phone calls automatically using the computer and present the caller with a list of options. The caller can then choose the appropriate option using the numbers on his phone key pad.

Using this feature you can implement a menu, for example: “for sales press 1, for support press 2 etc”. A digital receptionist is also known as an auto attendant.

You can configure different menu options and text for the menus based on which line the calls comes in, as well as based on whether the call is received within or outside office hours. This way you could have a different answer outside office hours and de-activate menu options accordingly.

Recording a menu prompt

Before you create your digital receptionist, you must write down the menu options you wish to offer the caller, think of a suitable text and record the announcement. A simple example would be “Welcome to Company XYZ, for sales press 1, for support press 2 or stay on the line for an operator”

Note: It is generally recommended to put the number the user should press after the option, i.e. “for sales, press 1”, rather than “press 1 for sales”. This is because the user will wait for the desired option and only then “register” what number to press

Once you have devised the text you can create the prompt and save the file in WAV format. For best performance, it is best to save the prompt in **PCM, 8 kHz, 16 bit, Mono** format. Windows Sound Recorder supports creation and conversion of this format.

Note: MP3 is not recommended because it requires more processing power to play and voice quality is not as good.

Recording the prompt in Windows Sound Recorder

To record the prompt in Windows Sound Recorder:

1. Start Windows Sound Recorder from the Start > Programs > Accessories > Entertainment group. This program should be installed by default on Windows.



Screenshot 13 - Windows Sound Recorder

2. Now click the record button to record your system prompt.
3. Now save the file using the 'Save As' command. This allows you to specify the format to save the file in. Click on the 'Format' button to change the format to PCM, 8 kHz, 16 bit. Enter a name for the prompt and click save.
4. When creating a digital receptionist menu you will be asked for the path to the file.

Note: You can also have your voice prompts professional recorded using a Voice Talent studio. Be sure to have them created in the specified format.

Creating a digital receptionist

You can create multiple digital receptionists and link them to a particular line. To create a digital receptionist:

1. In the 3CX Phone System Management Console menu, select on Add > Digital Receptionist.

Edit Digital Receptionist

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)

Name

Redirect To MS Exchange

Prompt

Menu options

key	key	Extension Number
0	Connect to Extension	100 Nick Galea
1	Connect to Extension	101 Richardson Bailey
2	Connect to Extension	102 Taylor Smith
3	Connect to Extension	103 Miller Cox
4	Connect to Extension	104 Bell Shaw
5	Connect to Extension	105 Thomas White
6	Connect to Extension	106 Johnson Jones
7	Connect to Extension	107 Adam Simpson
8	Connect to Extension	108 Peter Fisher
9	Repeat Prompt	
Timeout	60	End Call

Screenshot 14 - Configuring a digital receptionist

2. Specify a name and virtual extension number for the digital receptionist.
3. Now click on the browse button and specify the file that you previously recorded. The file will be copied into the 3CXPhone System\Data\lvr\prompts\ directory.
4. Specify the menu options. Select the appropriate key, and then select from the available actions. Then specify the extension number or virtual extension number (virtual extension number in the case of Ring Group, Call Queue or to another Digital receptionist)
5. The last option, Timeout, allows you to specify how long the system should wait for an input. If it receives no input, it will automatically perform this action. This is handy for callers who did not understand the menu or who do not have a DTMF capable phone. When ready, click OK to save the digital receptionist.

Allowing callers to dial a known extension directly

Whilst a digital receptionist prompt is playing, a caller can enter the extension number directly to be connected to an extension immediately. This allows callers who know their party's extension to avoid bothering the receptionist.

This option is enabled by default. If you wish to make use of this feature simply instruct your callers by explaining this in the voice prompt. For example:

"Welcome to Company XYZ, for sales press 1, for support press 2. If you know your parties extension number, you may enter it now"

Call By Name

You can also direct callers to the dial by name function. This allows them to find the person they wish to speak to by entering the first letters of the persons name on the phone dial pad. Requirements:

To call a user using the call-by-name mechanism you must:

1. Define a self-identification message for the user. Users without a self-identification message are not accessible via the call-by-name feature.
2. User can not have a last name with Unicode characters
3. Call-by-name menu dialog must be activated from a Digital Receptionist as an action on or timeout.

Self-identification Message

To record your self-identification message:

1. Go to your voice-mail menu (Default 999).
2. Go to the options menu ('9' key).
3. Press '5' key to record the self ID message.
4. Record your name only, i.e. 'Sarah Jones'

How it works

The Call-by-name feature uses the last name of the user and compares it with the input (that has been entered on the phone keypad). The following rules are used:

- The last name is converted to upper case.
- All symbols except [2-9] and [A-Z] are ignored.
- The following translations for symbols is used:

o 'ABC2' => '2'

o 'DEF3' => '3'

o 'GHI4' => '4'

o 'JKL5' => '5'

o 'MNO6' => '6'

o 'PQRS7' => '7'

o 'TUV8' => '8'

o 'WXYZ9' => '9'

You have to type a minimum of three digits ('0' – '9') to call to a user. Digits '0' and '1' are ignored, but can be used to call to users with short last names (for example, to access someone with the last name 'Li', you can type '540').

After the user has entered three digits, IVR queries the phone system database for matching users. If there are no matching users, it plays "extension not found". If there is only one matching user, the IVR plays "Please hold while I connect you to <user's self-identification prompt>" and redirects the call to the user. If there is more than one matching user, the IVR will wait for additional digits from the user for 2 seconds.

If IVR waits for additional digits (more than one matching user) and user presses any digit, the IVR will add this digit to the current input and check currently matching users. If there are no matching users, it will play "extension not found".

If the user does not input any more digits (2 seconds elapsed or '#' has been pressed) and more than one user is matched, then the IVR will play: "To call to Van Damme press 0. To call to Van Halen press 1. To exit press pound". In this example 'Van Damme' and 'Van Hallen' are self-identification prompts of the matching users.

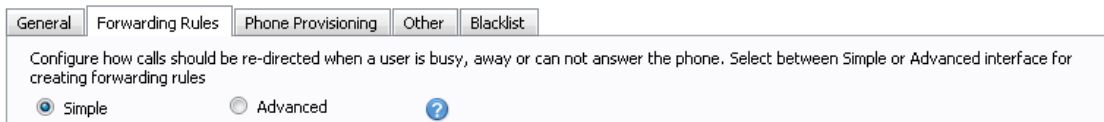
10. Call Forwarding rules

What are call forwarding rules?

The call forwarding rules are an important tool to deliver mobility to your users. Using the forwarding rules feature, you can configure exactly what should happen to calls when an extension is busy, a call remains unanswered or when an extension is unregistered. It is possible to configure this based on office hours, caller ID of caller and the type of call.

3CX Phone System has advanced call rule functionality and because delivering all this functionality can be overwhelming to many users, we have created 2 interfaces: A simple and an advanced call forwarding interface.

To configure the forwarding rules go to the extensions node and double-click on the extension you wish to configure. Then click on the 'Forwarding rules' tab.



Screenshot 15 - Selecting Simple or Advanced

Now switch to the Simple interface by clicking on the 'Simple' radio button.

Simple Call Forwarding Rules Interface

General Forwarding Rules Phone Provisioning Other Blacklist

Configure how calls should be re-directed when a user is busy, away or can not answer the phone. Select between Simple or Advanced interface for creating forwarding rules

Simple Advanced ?

Simple Advanced

No Answer

If call is not answered within seconds, then: ?

Send call to my voice mail

Send call to extension ?

Disconnect the call

Phone is busy, unregistered or set to away

If my phone is busy, unregistered or my status is set to away, then:

Send call to my voice mail

Send call to extension ?

Give busy tone

I want to be able to accept more than 1 call at the same time - uses Phone Status

Forward ALL calls / Out of Office

Forward ALL calls to:

My Voice Mail

Send call to extension ?

An external number or Skype ID

Offer option to reject call and divert to voice mail

Don't forward calls outside office hours (Send these to my voice mail)

Forward only external calls (Internal calls will be sent to my voice mail)

OK Cancel Apply

Screenshot 16 - The simple call forwarding rules interface

The Simple Call Forwarding rules interface allows you to easily setup call forwarding that will be sufficient for most users. You can configure where a call should be re-directed when a user does not answer, is on the phone or has set his/her user status to away. Additionally, if the user is out of the office, the Forward ALL section can be configured. The simple call forwarding interface will create the necessary Call forwarding rules. These can be viewed by switching to the advanced interface.

Configuring extension behavior on No Answer

In the section 'No Answer':

1. Specify the time in seconds that you want the system to wait before it considers the call unanswered.
2. Select one of the following options:
 - Send call to Voice Mail (of that extension)
 - Send call to extension
 - Disconnect

Configuring behavior when user is busy or his status is 'Away'

In the section 'Phone is busy':

1. Select one of the following options:

- Send call to Voice Mail (of that extension)
- Send call to extension
- Give busy tone

2. Now select whether you want to be able to receive a second call. By default, most IP phones accept a second incoming call when you are on the phone. If you want to redirect a call when you are already on the phone, tick the “I want to be able to accept more than 1 call at the same time” option.

Configuring Forwarding of ALL calls

Forward ALL calls / Out of Office

Forward ALL calls to:

- My Voice Mail
- Send call to extension
- An external number or Skype ID
- Offer option to reject call and divert to voice mail
- Don't forward calls outside office hours (Send these to my voice mail)
- Forward only external calls (Internal calls will be sent to my voice mail)

100 Nick Galea

three3cx

Screenshot 17 - Forwarding all calls

The Forward all calls option is typically used when a user is out of the office. This option will override ALL other rules and forward ALL calls to the configured destination. To configure it:

1. Tick the option ‘Forward ALL calls / Out of the office’
2. Select one of the following options
 - My Voice Mail (of that extension)
 - Send call to extension
 - An external number or Skype ID
3. If you selected forwarding to an external number, you can specify whether you wish to use the option ‘Offer option to reject call and divert to voice mail’ feature. This feature provides the option of the callee (the extension user) being able to reject the call and send it to the company voice mail system. This avoids the call being sent to the user’s mobile voice mailbox. How this works:
 - 3CX Phone System informs the caller “Please hold while we locate” “John” (user’s ID message) and puts the caller on hold (hold music).
 - 3CX now calls the specified external number and announces to the callee “You have a call from “0123456789” “Press 1 to take call, press 2 or hang-up to send to voicemail”. If the call is not answered within 30 seconds it goes to voicemail.

4. Select whether you want calls received outside of the company office hours to be redirected as well. If you do not select this option, these calls will be forwarded to the users voice mail.

5. Select whether only external calls should be forwarded. This option will send all internal calls to voice mail.

6. Click OK to save your configuration!

Note: forwarding calls to a Skype ID requires the Skype gateway to be installed and running.

Advanced Call Forwarding Interface

If you have more complex call forwarding requirements, use the advanced Call Forwarding Interface. With this interface you can set-up forwarding rules based on custom office hours, caller ID of caller, Call Type (Internal or External) and so on.

To switch to the Advanced Call Forwarding Interface, simply click the advanced radio button. You can switch back to the simple rules interface at any point. Your existing simple call forwarding rules will be listed automatically. However, note that when you switch back, your advanced call forwarding rules will be deleted and you have to re-configure simple call forwarding.

Edit Extension-100

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

General Forwarding Rules Phone Provisioning Other Blacklist

Configure how calls should be re-directed when a user is busy, away or can not answer the phone. Select between Simple or Advanced interface for creating forwarding rules

Simple Advanced

Simple Advanced

Forward Rules Settings:

Rule Type: No Answer
Hours: All Hours
Call Type: External Calls Only
Action: Forward to External Number using Voicer
Destination: 0035799558855

Add Rule Update Delete Move Up Move Down Clear

Rule Type	Hours	Call Type	Caller ID/DID	Action
No Answer	All Hours	External Calls Only		Forward to Voice mail 103

General Forwarding Options

Busy Detection Mechanism: Use Phone Status Use PBX Status
Continue Ringing for: 60 seconds

OK Cancel Apply

Screenshot 18 - Destination unreachable configuration

To create a forwarding rule in the advanced interface;

1. Specify a rule type:

- No Answer – Specify the time in seconds that you want the system to wait before it considers the call unanswered. This option can be set in the ‘Continue Ringing’ edit box at the bottom of the screen. This setting will apply to all ‘No Answer’ rules.
 - Phone Busy - You will have to specify whether you want the phone or the PBX to do the busy signaling. By default, most SIP phones do not give a busy signal back, but accept a second incoming call. If you want to redirect a call when an extension is busy, you will need to specify “Use PBX status” in the busy detection list box.
 - Phone not registered – If the SIP phone has not registered with 3CX Phone System (i.e. is turned off), then the phone is considered unregistered and the rule will kick in
 - Forward all Calls – This rule will forward ALL CALLS
 - Based on Caller ID – You can handle calls from specific callers by specifying their caller ID.
2. You can now specify a particular time during which hours this rule must apply. You can configure the rule to be valid all the time (All Hours), during or outside office hours (Relies on system wide office hours setting) or during specific hours (In this case you can configure the specific hours using the ‘Specific hours’ button)
 3. Specify the Call Type - Whether this rule should apply to all calls, or only internal or external calls.
 4. Specify the Action that the rule should take:
 - End Call
 - Forward to voice mail
 - Forward to extension –Specify the extension to forward to in the list box.
 - Forward to External number – In this case the system will forward the call to an outside number. Be sure to include the complete dial string including the prefix.
 - Forward to External number with the ‘option to reject call and divert to voice mail’ feature – This option forwards calls to an external number too, however it provides the option of the callee being able to reject the call and send it to the company voice mail system. How this works:
 - 3CX Phone System informs the caller “Please hold while we locate” “John” (user’s ID message) and puts the caller on hold (hold music).
 - 3CX now calls the specified external number and announces to the called person “You have a call from “0123456789” “Press 1 to take call, press 2 or hang-up to send to voicemail”. If the callee does not pick up within 30 seconds the call is sent to voice mail automatically.
 - Forward to Queue / ring group – Specify the virtual extension number of the ring group or Queue in the list box

- Forward to Digital Receptionist – Specify the virtual extension number of the digital receptionist in the list box.
5. Once you have filled out all necessary fields, click on the 'Add' button to create the new rule. You can order the priority of the rules by moving rules up or down. The first rule that meets the conditions will be triggered!

11. Adding DID numbers / Inbound Rules

Introduction

Many companies provide users and/or departments with “Direct or DID numbers”, which allow their contacts to call them directly, bypassing the receptionist. DID numbers are referred to as DDI numbers in the United Kingdom and as MSN numbers in Germany. Even if you have a digital receptionist, a direct line / number is often preferable because it’s more convenient for the caller.

Direct numbers can be easily implemented using DID numbers. DID numbers are provided by your VOIP provider or Phone Company and are virtual numbers assigned to your physical lines. Usually you are assigned a range of numbers, which is linked to an existing BRI/T1/E1. There will be an extra charge per number or per range, but this will be a fraction of the cost of adding physical lines. Enquire with your Phone Company or VOIP provider for more information.

DIDNumber	Port Identification	Virtual Line Number	During Office Hours	Out of Office Hours
*151	22446150	10001	809	202
*151	10003	10003	809	202
*152	10003	10003	117	203
*152	22446150	10001	117	203
*153	22446150	10001	118	202
*153	10003	10003	118	202
*476	22444032	10000	114	114
*476	10002	10002	114	114
*477	22444032	10000	112	112
*477	10002	10002	112	112
*478	22444032	10000	103	103
*478	10002	10002	103	103
*4033	22444032	10000	888	888
*4033	10002	10002	888	888
33361085095	0033179974750	70001	801	801

Screenshot 19 - Configuring DID's

DID numbers and 3CX Phone System – How it works

DID numbers “play well” with 3CX Phone System, or indeed with any SIP based IP PBX. Acting according to the SIP standard, 3CX Phone System expects the DID number, i.e. the intended destination, to be in the To: field of the SIP invite request. Most VOIP providers and VOIP Gateways will do this by default. Then all you need to do in the 3CX Management Console is to configure calls made to that particular DID number to go to a particular extension, digital receptionist or other destination.

Adding DID's

Add DID

Route calls to DID/DDI numbers directly to an extension

Route calls for this DID/DDI Number
 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 dialled number of +35722444032 in the "to" field

DID/DDI Name:

DID/DDI number/mask:

Available ports:

- Select all
- 22444032[ISDNBR4554]
- 22446150[ISDNBR4554]

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 email of extension:

Screenshot 20 - Selecting where to route calls to this DID

To add a DID;

1. Click on the 'Create DID' button in the 3CX Management Console in the toolbar.
2. Enter a name for the DID (for example sales).

Note: The DID name can be pre-pended or appended to the Caller ID so as to identify on which number a caller has called. You can enable this from the Settings > General > Global options page under 'Append/Prepend name to caller ID'

3. Now enter the DID number as it will appear in the SIP "to" header. 3CX Phone System will match the number inserted in this field with the "to" header, starting from the last part of the received string, thus avoiding any differences in the format of the number. For example, if you are based in the UK and your DID number is 0845-2304024, then you can enter the number 2304024. This will match any DID number inserted in the "To" field ending with these numbers, including +448452304024, 08452304024, 00448452304024, and, of course, 2304024. Of course it is best to check with your VoIP provider or Phone Company in what format the DID number will be inserted just for reference.

4. Now select for which ports you wish to add this DID. If the DID number is associated with multiple ISDN ports, then you must select each. An inbound rule will be created for each port that you select.

5. Now specify where you wish to direct calls made to this DID:

- End Call
- Connection to extension
- Connect to Queue/Ring Group
- Connect to Digital receptionist
- Voicemail box for extension
- Forward to outside number

- Send fax to email of extension

6. You can specify that an incoming call is routed differently if it is received outside office hours. De-select the 'Same as during office hours' option to specify a different route.

7. Click OK to create the DID / Inbound rule. The newly created DID's will be listed as inbound rules.

Using DID's with a VoIP provider account

If your VoIP provider has supplied you with DID's, AND inbound call identification is based on destination/dialled number, it will be necessary to configure the Source Identification rules for the VoIP Provider. Please refer to the chapter on 'Adding lines hosted by a VoIP provider' for more information.

Troubleshooting DID lines

If you have created DID lines, but calls are not being forwarded, do the following:

1. Go to the Server Activity log node in the 3CX management Console. The Server Status screen lists current server activity and logs calls that are being received and for which number they were received.
2. Call the DID number that you configured, and monitor the Server Status log. You will see a line appearing something like:
Incoming call from 1000 to <sip:789456123@3CXPhone System>
where "1000" is the internal number of the line configured to receive calls from the VOIP Gateway or VOIP Provider and *<sip:789456123@3CXPhone System>* is the content of the "To" header of the INVITE, i.e. the intended recipient.
3. Now analyze the "To" header carefully and ensure that the DID number you have dialled is present in the "To" header: *<sip:789456123@3CXPhone System>*.
4. If the DID number is not present in the "To" header, you will have to check the documentation of your VOIP Gateway to find how you can configure it to insert the DID number into the "To" field. In the case of a VOIP Provider, contact your VOIP provider for more information.
5. If the DID number is present in the "To" header, check whether the string listed in the "To" field matches the string you configured in the rule. Adjust accordingly and try again.

12. Ring groups, Call Queues, Paging / Intercom

Ring Groups

A ring group allows you to direct calls on particular numbers to a group of extensions. For example, you could define a group of 3 sales people, and have the general sales number ring on all 3 extensions at the same time or after each other. When you create a ring group, you also assign it a virtual extension number. This will be the number used by the phone system to “address” the ring group.

Edit Ring Group

Ring groups allow more than one phone to ring at the same time or in a sequence

General

Enter the Ring Group details. The phones will ring until one of them is answered or until the timeout is reached.

Virtual machine number (can not be in use as extension)

Name

Ring Strategy

Ring Time (Seconds)

Use Multicast for Paging

Ring Group Members

Select which extensions are a member of this Ring Group.

Extensions

100 Nick Galea

Add >

< Remove

Members

101 Joe Bloggs

102 Sandro Walt

103 John Smith

Up

Down

Destination if no answer

Select a destination for this call if the call goes unanswered.

End Call

Connect to Extension

Connect to Queue / Ring Group

Connect to Digital Receptionist

Voicemail box for Extension

Forward to Outside Number

OK Cancel Apply

Screenshot 16 - Adding a Ring Group

To add a ring group:

1. In the 3CX Phone Management console menu, select **Add > Ring group**.
2. Now enter the ring group options:
 - Virtual extension number – Specify an extension number which will be reserved for this ring group. Note that this extension number will be automatically created. Do not specify an existing extension number.
 - Name – Enter a friendly name for the ring group
 - Ring strategy – Select the appropriate ring strategy for this ring group:
 - Hunt – this will start ringing on the first extension, then the second etc.
 - Ring all – all phones will ring at the same time
 - Paging – this will page all extensions part of the group (see next section)

- Ring time – Specify how long the phones should ring for.

3. In the section 'Ring group members' specify the extensions that should be part of this ring group. Simply click on the extensions and click on the -> to make them a member. Move the extensions up or down to configure the priority of an extension.

4. In the section 'Destination if no answer', you can define what should happen if the call does not get answered by the ring group.

Paging / Intercom (Paid editions only)

Paging allows some one to ring a group of extensions and make an announcement via the phone speaker. The called party will not need to pick up the handset. The connection will be one way audio.

The intercom feature allows a phone system user to make an announcement to a single extension. In this scenario the audio is two way, and the called party can respond immediately without picking up the handset.

Both paging and intercom features require a phone that supports intercom and that is configured to allow it. See the phone configuration guides for more information.

To add a paging group:

1. Click on the **Add > Ring Group** menu option to bring up the 'Add Ring Group' page.
2. Now enter the ring group options:
 - Virtual extension number – Specify an extension number which will be used for this paging group.
 - Name – Enter a friendly name for the ring group
 - Ring strategy – Select the Paging ring strategy

3. If you have phones that support multi cast, and you have a very large installation with specialized requirements, you can enable the Multi cast option. For most installations this option is not required. See the wiki for more information.

4. In the section 'Ring group members' specify the extensions that should be part of this paging ring group. Simply click on the extensions and click on the -> to make them a member.

Note: The 'Ring time' and 'Destination if no answer' options will be ignored, since they are not relevant for paging.

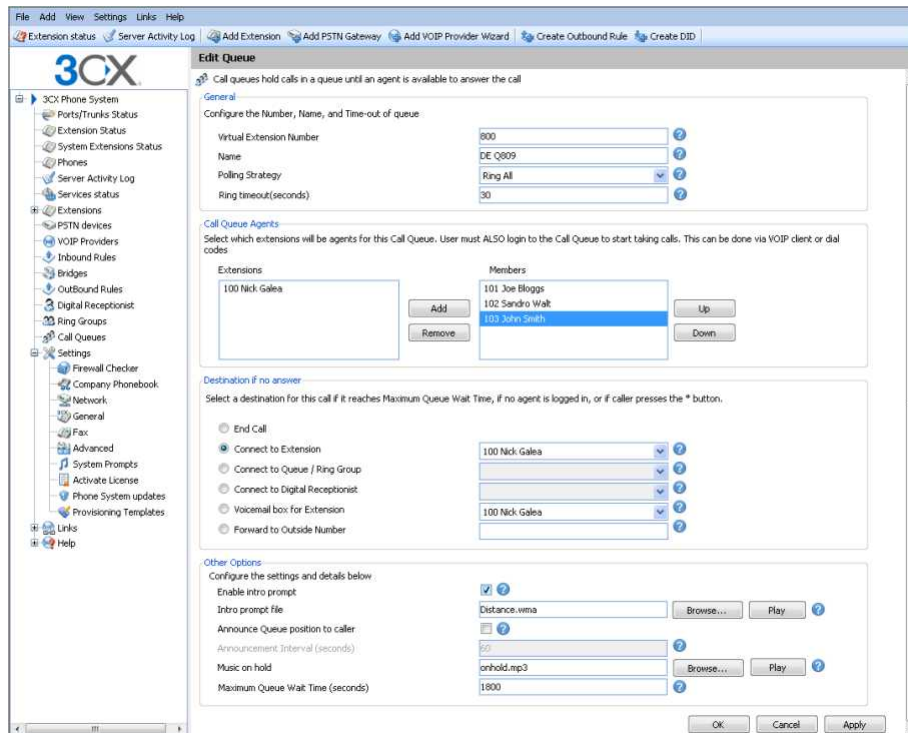
To call a user via the intercom function:

Prefix the extension you wish to call with ***9**, followed by the extension. For example to make an intercom call to extension 100 you should dial:

'*9100'

Call Queues (Paid editions only)

Call Queues allow calls to be queued whilst agents (members of a call queue) answer calls. Calls do not go unanswered but are queued until an agent is available to take the call. For example, you can define a group of 3 sales people, and have the general sales number route to a sales call queue. If all 3 sales people are busy, callers will be kept on hold until the next sales person is free. When you create a call queue, you also assign it a virtual extension number. This will be the number used by the phone system to “address” the Call Queue.



Screenshot 17 - Adding a Call Queue

To add a Call Queue:

1. Click on the **Add > Call Queue** menu option to bring up the 'Add Call Queue' page.
2. Now enter the call queue options:
 - Virtual extension number – Optionally change the suggested virtual extension number. Note that this extension number will be automatically created. Do not specify an existing extension number.
 - Name – Enter a friendly name for the ring group
 - Polling strategy – This option allows you to choose how calls should be distributed to the agents:
 - Hunt random start – 3CX will randomly choose an agent to distribute the call to. This will evenly distribute the calls to each of the agents
 - Ring All – the phones of ALL the agents will ring

- Hunt – 3CX will distribute the call according to the order specified in the Queue members section. All calls will go to the first agent first, and only if this one is busy, it will go to the next agent. This strategy can be used to setup skills based routing, by ordering the agents according to their skills.
- Ring time – Indicate the time out, i.e. for how long the phone should keep ringing before it considers the call unanswered by that agent.

3. In the section 'Call Queue agents' specify the extensions that should be part of this Call queue. Simply click on the extensions and click on the -> to make them a member. Move the extensions up or down to configure the priority of an extension.

In addition to being a member, an extension/user must also login to start answering calls routed to this call queue. Users can login to a call queue using the login button in the VOIP Client.

4. In the section 'Destination if no answer', you can define what should happen if the call does not get answered by an agent. If no agent is logged into the queue, this option gets triggered immediately. In addition, this option gets triggered if the user presses the '*' button on his phone. This allows you to give callers an option to exit out of the queue and leave a message.

5. In the section 'Other options', you can specify a custom introduction prompt and a custom music on hold file. You can also decide whether you wish to announce a caller's position in the queue and what the maximum wait time should be.

13. Call conferencing

Introduction

Note: This feature is not present in the free edition of 3CX Phone System

Call conferencing allows you to easily setup up to eight conference calls that can allow a total of 32 callers (license permitting). In other words the 32 caller limit is for all conferences, not per conference. Note that a conference with 32 participants will require a powerful machine.

Although many conference call services exist, it's often easier and cheaper to host your own audio conferences.

Conferences can be set-up ad hoc, without the need to reserve a conference room. This has been done to simplify the set-up of conference calls.

Enabling conferencing

The conferencing service does not require much configuration – you just need to specify the default conference extensions. To do this:

1. In the 3CX Management console, open the Settings > Advanced node and click on the Conferencing tab.
2. Now specify the conferencing extension number. This is the number that users must call to setup a conference.
3. Specify the maximum number of conferences you wish to support. By default 4 conferences can be held at a time.

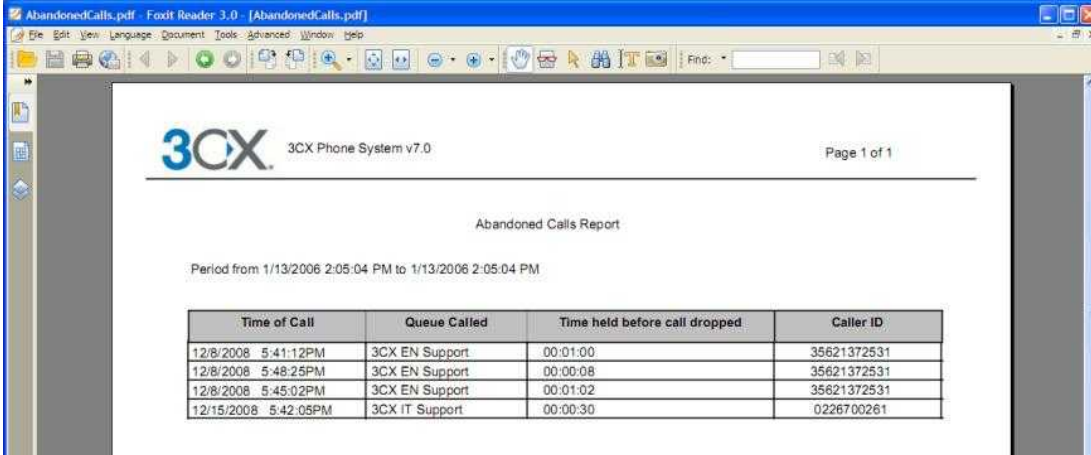
Creating a call conference

To setup a call conference;

1. Dial the conference extension number, by default this is 700
2. You will be requested to enter a conference ID. This can be any number, for example '100'. All callers that wish to join the conference must use the same conference ID!
3. If you are the first participant, the conference system will ask you to confirm creation of the conference. Press * to confirm or # to cancel.
4. You will be asked to speak your name after the beep and press a button to continue.
5. You will now enter the conference. If you are the first caller, you will hear music on hold, whilst you wait for the other callers. As soon as another caller joins, his name will be announced.

14. Generating Call Reports

Introduction



Time of Call	Queue Called	Time held before call dropped	Caller ID
12/8/2008 5:41:12PM	3CX EN Support	00:01:00	35621372531
12/8/2008 5:48:25PM	3CX EN Support	00:00:08	35621372531
12/8/2008 5:45:02PM	3CX EN Support	00:01:02	35621372531
12/15/2008 5:42:05PM	3CX IT Support	00:00:30	0226700261

Screenshot 21 - Call Reports

3CX provides a number of reports via its 3CX Call Reporter utility. This utility can be started from the 3CX Phone System program group.

Reports available

After you have started 3CX Call Reporter, you can access the following reports from the reports menu at the top:

- Call Logs Report
- Call Statistics
- Queue Statistics
- Abandoned Calls
- Agent statistics
- Ring Group Statistics

15. Connecting 3CX Phone Systems

Introduction

Note: This feature is not present in the free edition of 3CX Phone System

You can connect 2 separate 3CX Phone Systems or a 3CX Phone System with another SIP phone system using a bridge, allowing you to make calls between branch offices using your internet connection – and thus at no charge.

The “bridge” will be assigned a prefix, which users must dial to access the other 3CX Phone System or SIP phone system. This prefix must be followed by the extension number of the user they wish to reach on the other 3CX Phone System. For example, if you assign the prefix “2” to a bridge with another office, and within that office you want to dial someone who has extension number 105 on that phone system, you would dial 2105 to reach that person directly.

Creating a bridge

To create a bridge:

1. Click on the **Add > Bridge** menu option to bring up the ‘Add Bridge’ page.
2. Enter a name for the bridge and assign a virtual extension number. A bridge must be assigned a virtual extension number so it can be addressed by the phone system. Accept the default or choose another extension that is free.
3. Now select the Type of bridge:
 - Master (Direct - UDP) - If you select ‘Master’, the other PBX must register with this system. In Direct - UDP mode, all traffic will be sent via UDP and will use multiple ports.
 - Master (Tunnel - TCP) - If you select ‘Master’, then the other PBX must register with this system. The tunnel option allows all SIP and RTP traffic to be sent via a single TCP port. The 3CX Tunnel vastly simplifies firewall configuration, although it can not provide the same quality as a direct connection. Also, the tunnel option can only be used against another 3CX Phone System.
 - Slave (Direct - UDP) – This system will register with the remote system using direct SIP.
 - Slave (Tunnel - TCP) – This system will register with the remote system using the 3CX tunnel.

Type of Bridge
Configure whether this bridge should be master, slave or use the tunnel. As slave will register with the master using a direct UDP connection. The tunnel option allows you to "tunnel" SIP traffic via a single TCP port.

Type of 3CX Bridge ⓘ

Bridge - Master (Tunnel(TCP or UDP))
The other PBX must register with this one. Enter the credentials that the other PBX (slave) should use to authenticate with this PBX.

Authentication ID ⓘ

Authentication Password ⓘ

Screenshot 22 - Creating a master bridge

4. If you selected 'Master (Direct-UDP)' then all you need to do is enter the authentication details which must be used by the slave to register with this 3CX Phone System. **The authentication ID must be UNIQUE on this phone system and these same credentials must be used by the slave bridge.**

5. If you selected 'Master (Tunnel-TCP)' then you must enter

- Authentication ID and Password –**The authentication ID must be UNIQUE on the phone system and these same credentials must be used by the slave bridge.**
- Remote end of the tunnel – Enter the public IP of the REMOTE 3CX Phone System machine.
- Enter the remote port of the 3CX Tunnel (by default 5090)
- Enter the port of the Local end of Tunnel. For the first bridge connection it is 5081 (5080 is used for external extensions). The port will be incremented by 1 for each bridge you create that uses the tunnel.

Type of Bridge
Configure whether this bridge should be master, slave or use the tunnel. As slave will register with the master using a direct UDP connection. The tunnel option allows you to "tunnel" SIP traffic via a single TCP port.

Type of 3CX Bridge ⓘ

Bridge - Slave (Direct(UDP))
This PBX must register with the other PBX. Enter the IP of the remote PBX and the credentials to authenticate with that PBX.

Remote end of Tunnel (Public IP of Remote 3CX Phone System) ⓘ

Authentication ID ⓘ

Authentication Password ⓘ

Time between registration attempts(in seconds) ⓘ

Port ⓘ

Screenshot 23 - Configuring a slave bridge

6. If you selected to create a slave bridge using direct UDP, then you must enter:

- Public IP of the remote 3CX Phone System
- SIP Port of the remote 3CX Phone System

- Authentication ID and Password – The credentials with which this side of the bridge must authenticate with the ‘Master’. These must of course match the credentials entered on the master.

Type of Bridge
Configure whether this bridge should be master, slave or use the tunnel. As slave will register with the master using a direct UDP connection. The tunnel option allows you to "tunnel" SIP traffic via a single TCP port.

Type of 3CX Bridge: Tunnel Connection ⓘ

Bridge - Tunnel(TCP)
The tunnel option tunnels all SIP and RTP traffic via a single TCP port, vastly simplifying firewall configuration. This option can only used with another 3CX Phone System and a slight quality degradation might be noticeable.

Local IP or Hostname of remote 3CX Phone System	<input type="text"/>	ⓘ	Port	<input type="text" value="5060"/>	ⓘ
Remote end of Tunnel (Public IP of Remote 3CX Phone System)	<input type="text"/>	ⓘ	Port	<input type="text" value="5090"/>	ⓘ
Local end of Tunnel	<input type="text" value="localhost"/>	ⓘ	Port	<input type="text" value="5080"/>	ⓘ
Authentication ID	<input type="text" value="10004"/>	ⓘ			
Authentication Password	<input type="text" value="10004"/>	ⓘ			
Time between registration attempts(in seconds)	<input type="text" value="60"/>	ⓘ			

Screenshot 21 - Slave bridge using 3CX Tunnel

7. If you have selected to create a slave bridge using a 3CX Tunnel, then you must enter

- Local IP or hostname of remote 3CX Phone System and port
- Remote end of the tunnel – in most cases the tunnel will be running on the 3CX Phone System machine, in which case you need to enter the public IP of the remote 3CX Phone System machine. Enter the remote tunnel port of the 3CX Tunnel (by default 5090)
- Enter the Local end of Tunnel. Leave the default ‘Localhost’ and 5081 to use the Tunnel installed by default with 3CX Phone System. The port will be incremented with each bridge that you create that uses the tunnel.
- Authentication ID and Password – Enter the credentials with which this side of the bridge must authenticate with the ‘Master’. These must of course match the credentials entered on the master.

Note: You will have to open port 5090 on the firewall behind which the ‘Master’ 3CX Phone System resides.

8. Select provider capabilities – Enable all options if the remote system is a 3CX Phone System.

9. Select which codecs you wish to use. You can select GSM, Speex, Ilbc or G729 to save bandwidth. Note you must turn on ‘PBX delivers audio’ to ‘enforce this codec’.

10. Now you must assign a prefix that users must dial to access the bridge. Click Next to create the bridge.

11. After it is created it, you can edit bridge options by going to the bridges node. You can edit:

- In the section other options you can define the maximum number of simultaneous calls you will allow for this bridge and whether outgoing or incoming lines will be allowed.
- Configure what to do with calls coming in via the bridge outside office hours

10. Now you must go to the Management Console of the other 3CX Phone System and set-up the opposite end of the bridge, i.e. either a 'Master' or a 'Slave'. You must use the same authentication credentials!

Calling a party on the other 3CX Phone System

To dial a party at the connected 3CX Phone system, you must dial the assigned prefix, plus the extension number of the person you wish to call.

16. Using the tunnel for remote extensions

Introduction

A powerful benefit of a software based IP PBX is the ability to support remote extensions, i.e. employees using their extension from home or a satellite office. This gives tremendous flexibility to employees and truly delivers mobility, because employees working from home or at remote offices can be seamlessly integrated with head office. They can be a member of call queues and can use the 3CX assistant to see presence of other users.

Traditionally it has been quite difficult to configure remote SIP extensions because of NAT and firewall configuration issues. 3CX has resolved this problem by developing a unique Tunnel service, which tunnels all voice traffic over a single TCP/UDP port and requires only a single port to be opened on the server.

The tunnel can carry SIP, RTP traffic, HTTP traffic for the MyPhone user portal as well as 3CX Assistant traffic. 3CX Phone System can handle multiple tunnel connections from different locations, and if you want you can proxy connections from several phones in one remote location via a single tunnel connection.

With the 3CX Tunnel it easy to setup a remote extension!

How it works



Figure 4 - How a tunnel works

The picture above demonstrates how a tunnel works. In this example, the 3CX Phone System is on IP Address 10.0.0.181, and listens on TCP port 5090 (by default) for incoming Tunnel traffic. We must set up a single Port Forwarding rule on the Modem or NAT/Firewall Device, telling it that all incoming TCP traffic received on port 5090 should be delivered to LAN IP Address 10.0.0.181.

The remote setup is shown on the left hand side of the cloud. In this example, the Machine IP Address 192.168.0.2 has the 3CX VoIP Phone installed. We will need to tell the VoIP Phone the public IP address of the PBX Server (which in this case is 213.165.190.51), and also the private IP address of the PBX Server (which in this case is 10.0.0.181).

Since the 3CX VoIP Phone will by default use the standard port numbers used by 3CX Phone System, typically no further configuration will be necessary!

Configuring the tunnel

We will now use the above example in 'How it works' to configure a tunnel connection.

Step 1 – Configure the PBX

1. In 3CX Management Console, go to the Settings > Network > 3CX Tunnel page.

- Set the Tunnel Password to “3cx”
- Set the Local IP to the Local IP Address of the NIC which will be receiving tunnel connections. If the PBX has only one NIC, then there will be no need to set this field. In our example this is 10.0.0.181
- Set the Tunnel Listening Port to the port which will be receiving tunnel connections. The default value is 5090.
- Click the “OK” button. The Tunnel service will be restarted automatically.

Step 2 – Configure the Firewall

The Tunnel protocol is designed to eliminate NAT traversal problems and reduce Firewall configuration work to a minimum. There is only one Firewall setting that needs to be made – we must forward the TCP Tunnel port (set by default to 5090) to the PBX.

Home Wizard Wireless Settings Firewall Advanced Settings Toolbox Choose your language WL-183

Entries in this table allow you to automatically redirect common network services to a specific PC behind the NAT firewall. These settings are only necessary if you wish to host some sort of server like a web server or mail server on the local network.

Enable Port Forwarding

Local IP	Type	Port range	Comment
10.0.0.181	TCP	5090 - 5090	3CX Tunnel

Add Reset

Current Port Forwarding Table:

NO.	Local IP	Type	Port range	Comment	Select

Delete Selected Delete All Reset

Apply Cancel

Wireless Network Broadband Router 300N

Screenshot 24 - Configure a port forward rule

The above picture shows configuration for a Sitecom WL-183 WAN-to-LAN router, but most routers will provide similar functionality. In your firewall:

1. Enable Port Forwarding
2. Specify the PBX's Local IP Address (which we had set previously to 10.0.0.181)
3. Set the Type to “TCP”
4. Set the Port Range to be from 5090 to 5090 (only one port)
5. Set the Comment field to “3CX Tunnel”
6. Click on the “Add” button followed by the “Apply” button

Your firewall configuration is now done!

Step 3 Configure 3CXPhone

The screenshot shows the 'Connection settings' dialog box with the following fields and values:

- Profile:** Tunnel Connection (dropdown menu)
- Credentials:**
 - Extension: 100
 - ID: 100
 - Password: ***
- My location:**
 - I am in the office - local IP
 - I am out of the office - extern IP (212.165.189.54 of PBX)
- Tunnel:**
 - Use tunnel
 - Local IP of remote PBX: 10.0.0.158
 - Tunnel password: ***
 - Port: 5090

Screenshot 25 - Configuring 3CXPhone

Now we will configure the 3CX VoIP Phone to communicate with the 3CX Phone System. When the phone starts up the first time, you are presented with a blank profile.

1. Rename the profile to “Tunnel Connection” – alternatively you can create a new profile to do this.
2. Set the “Extension” field to your extension number – in this case Extension “100”
3. Set the “ID” field to the Authentication ID for your Extension – in this case “id100”
4. Set the “Password” field to the Authentication Password for your extension – in this case “pw100”
5. In the “My Location” section, set the option to “I am out of the office” – and set the External IP Address of the PBX to “213.165.190.51”
6. In the “Tunnel” section, enable the “Use tunnel” checkbox
7. Set the “Local IP of remote PBX” field to “10.0.0.181”
8. Set the “Tunnel Password” field to “3cx”
9. Click on the “OK” button

Your tunnel connection is now configured!

More information

To learn more about tunnel connections and to learn how to troubleshoot remote extensions, you can view the video tutorial on this subject in the 3CX Online training area:

<http://training.3cx.com/extensions/the-3cx-tunnel-and-configuring-3cx-voip-phone-as-a-tunneled-external-extension>

Using the tunnel with IP hard phones

It is also possible to use the tunnel with an IP phone. More information and the download of this utility can be found in this wiki article:

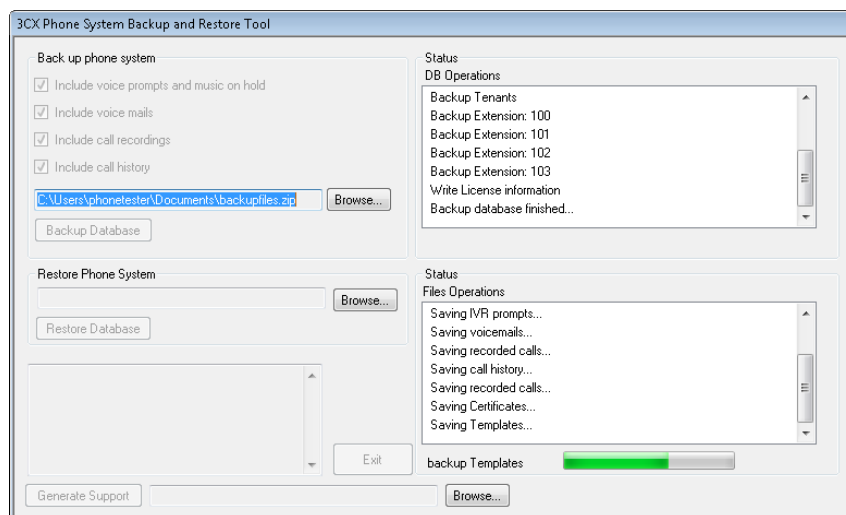
<http://wiki.3cx.com/documentation/networking/sip-proxy-manager>

17. Backup and Restore

Introduction

3CX Phone System includes a convenient backup and restore utility, that allows you to create a complete backup of your phone system configuration and data to a file. To backup data, run the Backup and Restore utility located in the 3CX Phone System Program group.

To restore phone system data, locate the backup of the phone system data and click 'Restore'. Note that the current configuration will be **OVERWRITTEN** with the backup data, and any changes that you have made since performing the backup will be lost. You must also close the management console **BEFORE** making a restore.



Screenshot 26 - Backup & Restore utility

Scheduling a backup

Using the Windows scheduler you can easily schedule a daily phone system backup. To do this:

1. Go to Start > Accessories > System tools > Schedule Tasks
2. Double-click on 'Add Scheduled task'
3. Browse to the 3cxbackup program, 3cxbackup.exe
4. Specify schedule and account to use.
5. After it is created, you have to modify the schedule to include the command line parameters that you need:
 - Hidden - runs the process hidden so it will close automatically after completion
 - Backup - will backup the database
 - Restore - will restore the database

- Filepath is the location of the database to restore from or backup to
- Options
 - /callhist - will include the call history database
 - /voiceprompts - will include system prompts
 - /voicemails – will include voice mails
 - /callrecordings - will include call recordings.
 - /exit – to exit the utility after backup is done.

Example: Complete hidden backup with exit

```
3CXBackup.exe hidden backup c:\backup.zip /callrecordings /voicemails /voiceprompts /callhist /exit
```


18. Fax server

Introduction

Note: This feature is not present in the free edition of 3CX Phone System

3CX Phone System includes a fax server that allows the whole network to receive faxes. The 3CX fax server is based on the T38 standard and requires a compatible T38 VOIP gateway or provider. Note that to use fax you must use a supported gateway and it must be configured according to our configuration guides, so that fax reception is enabled. It is also possible to use a VoIP provider that supports T38, however the quality of the fax implementation between VoIP providers varies and can therefore not be guaranteed.

Fax receiving configuration

The receiving of faxes is handled by the 3CX Fax Server service. You must configure a line or a DID to be dedicated to fax, so that all fax calls are forwarded to the 3CX Fax Server. The 3CX Fax server will then receive the fax, convert it to PDF and then email the fax to the configured email address.

Route calls for this DID/DDI Number to

Enter a DID or string to look for in the SIP "to" field, starting at the end of the string. Use wildcards (*) to match any digit for that entry. For example, entries 22444032 OR 2244403* will both match calls with +35722444032 in the to field

DID/DDI number/mask: for port number: 10001

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 email of extension:

Same as Out of Office hours

OK Cancel Apply

Screenshot 27 - Configuring a port or DID to receive a fax

To do this:

1. In the Management console, select the port or DID which will be dedicated to receiving faxes.
2. Select 'Send fax to email of extension'
3. Select to which email address you want to email the received fax. If you select "Default FAX Destination", the fax will be sent to the email address configured for the virtual fax extension number. Alternatively you can forward faxes to the email address configured for a particular extension. This allows you to create multiple DID rules so as to give people a personal fax extension. You can configure the default email address from the settings > fax node.

Changing the Fax Server Credentials

The Fax server operates as a virtual extension and logs into the SIP server just as an extension would. It uses an authentication ID and password which is set to 888 by default. To change the authentication ID and password:

1. In the 3CX management console, go to the Settings > Fax node
2. In the 'Fax Server Extension Number' field, specify the fax extension number. Any call forwarded to this extension will be assumed to be a fax and receive a fax tone.
3. Specify the Fax Server Authentication ID and Password – These credentials will be used by the fax server to login to the SIP server.

Please note that you must restart the fax service for changes to take effect.

19. The Phonebook

Introduction

The phonebook feature allows you to easily publish a company wide phonebook. Used in tandem with a personal phonebook, it allows users to quickly launch calls without wasting time finding a person's number and subsequently entering it in the phone.

3CX Phone System supports a company and a personal phonebook. The Company phone book is company wide and is managed from the management console. The personal phonebook is only available to a particular extension and is managed from the MyPhone page of that extension.

The company phonebook is also published to a directory in a format that SNOM, Grandstream, Aastra and Polycom phones can download. These phones can then show the same phonebook on their display.

Company phonebook

First Name	Last Name	Phone
Alex	Mitchell	98745566
Edwards	Hall	99874552
Jackson	Lewis	97888552
John	Taylor	79885522
Thomas	Collins	97778855
Walker	Wright	79988553

Screenshot 28 - The company phonebook

To manage the company phonebook, go to the Settings > Phonebook node. Click 'Add' to add an entry.

Importing Phonebook entries

You can import phonebook entries from a CSV file. Each entry should be on a new line, and the fields separated by a comma as follows:

First name, Last name, Phone number

Using the Phonebook

To use the phonebook, users enter a name or part of the name in the 'Make Call' Dialog in the 3CX Assistant. 3CX Assistant will automatically resolve the name or part of the name to a phonebook entry. To launch a call, the user just selects the name and clicks the 'Call' button.

20. Monitoring your Phone System

Introduction

3CX Phone System is easy to monitor for any Windows administrator, since it behaves just like any other Windows Server application. You can monitor 3CX Phone System using your favourite network monitoring solution, such as for example ActiveXperts or Microsoft Operations manager.

Things to monitor

Systems extensions status

System Extensions Status				
✘ Disconnect Call				
Status	Extension	Type	IN/OUT	
● Registered (idle)	*0	ParkExtension		
● Registered (idle)	*1	ParkExtension		
● Registered (idle)	200	IVR		
● Registered (idle)	201	IVR		
● Registered (idle)	202	IVR		
● Registered (idle)	203	IVR		
● Registered (idle)	700	ConferencePlaceExtension		
● Registered (idle)	701	ConferencePlaceExtension		
● Registered (idle)	702	ConferencePlaceExtension		
● Registered (idle)	703	ConferencePlaceExtension		
● Registered (idle)	704	ConferencePlaceExtension		
● Registered (idle)	800	RingGroup		
● Registered (idle)	801	IVR		
● Registered (idle)	802	Queue		
● Registered (idle)	803	RingGroup		
● Registered (idle)	804	RingGroup		
● Registered (idle)	805	IVR		
● Registered (idle)	806	IVR		
● Registered (idle)	807	IVR		
● Not Registered	808	IVR		
● Registered (idle)	809	Queue		
● Registered (idle)	888	FaxExtension		
● Registered (idle)	999	SpecialMenu		

Screenshot 29 - Monitoring System extensions

3CX Phone System uses system extensions for services such as IVR, Queue, Fax, Parking and so on. Using the System extensions node in the 3CX Management Console you can quickly monitor if all these system extensions are working and registered correctly.

3CX services

A good first check is to monitor all 3CX services are running. You can view all 3CX services from the services node in the 3CX Management Console. Any network monitoring package can monitor windows services remotely.

Windows events

The following events are posted and can be monitored using your network monitoring package:

[CM104001]: -- "registration failed because specified destination could not be resolved or doesn't respond"

This event occurs when the remote VOIP provider could not be contacted. This could occur when your internet connection is down or the specified IP or FQDN for the VoIP provider is incorrect or down.

[CM104002]: -- destination has responded with error code.

This event occurs when the VOIP provider responds with an error, for example, it could be that your account has been disabled

Monitor IP's of gateways and phone system

Additionally, you should create checks that regularly check the IP of any VoIP gateway as well as the phone system to ensure that they are up and running

21. Troubleshooting

Introduction

If you have problems configuring 3CX Phone System, you can access the following sources of information for help:

- The manual – most issues can be solved by reading the manual.
- The 3CX Wiki – <http://wiki.3cx.com/>
- The Phone configuration guides – <http://www.3cx.com/sip-phones/index.html>
- The Gateway configuration guides - <http://www.3cx.com/voip-gateways/index.html>
- Follow our online training course – <http://training.3cx.com>
- The Support forum – <http://www.3cx.com/forums/>
- Our support system (requires support package)

Manual

The manual describes the installation process in detail. Many of the questions we receive in support are clearly documented in the manual. You can always find the latest version of the manual here:

<http://www.3cx.com/support/index.html>

Configuration guides

Be sure to follow the configuration guides for your gateway or sip phone:

- SIP Phone configuration guides – <http://www.3cx.com/sip-phones/index.html>
- VOIP Gateway configuration guides - <http://www.3cx.com/voip-gateways/index.html>

Wiki

3CX maintains a Wiki, which includes answers to the most common problems:

<http://wiki.3cx.com>

Request support via our support system

If you have purchased a support package from 3CX, you can contact the 3CX support department via the support system. Login details would have been provided to you by email.

When requesting support, include the 'Support info' data. 3CX Phone System can automatically generate a file which includes all relevant support information. **NO**

PASSWORDS TO PHONES OR VOIP PROVIDERS WILL BE INCLUDED. The data will NOT be sent automatically. You will be prompted for a location to save the data, so you can check what data will be sent to us before you send it to us.

To generate the support info file:

1. In the 3CX Phone System Program Group, start the 'Backup and Restore' tool.
2. Click on the button 'Browse' next to 'Generate Support'.
3. You will be prompted for a location to save the data. Enter the file name to be generated. You can review the data that will be sent prior to sending it to us.
4. Login to the 3CX support system, and attach the information to your support request.
5. Include a detailed problem description. It should clearly indicate what the exact problem is, and when it occurs. Mention what hardware or VoIP provider you are using with 3CX Phone System.

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