Manual

3CX Phone System for Windows

Version 9.0

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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 then 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.

Allows easy roaming of users:
Calls can be diverted anywhere in the world because of the SIP protocol characteristics

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.

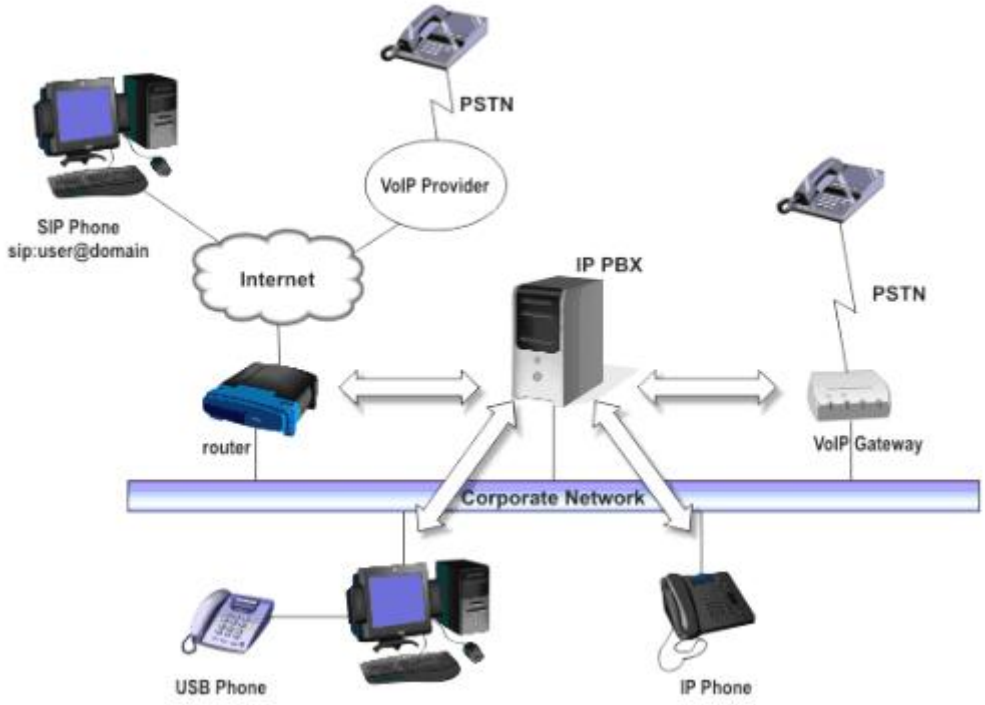


Figure 1 - VOIP Phone System Overview

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

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 V9?

All versions

- Video support using 3CXphone, X-lite and Yealink videophones

- G722 codec support
- New inbuilt high performance web server (Abyss) which can scale to any installation size. IIS can still be used if required.
- Ability to provision 3CXPhone and 3CX Assistant
- Additional node to quickly view all remote extensions
- Additional status information on VoIP provider trunks and bridges
- Ability to specify a mobile number for a user
- Improved call forwarding rules design
- Improved performance of the media server
- System wide black list
- Ability to specify up to 5 music on hold files
- Ability to send a 302 diversion header to forward a call to another number.
- Ability to import only new users from Active Directory since last import

Small Business, Pro and Enterprise editions

- New SLA like parking functionality to allow for easy parking and un-parking of calls via BLF lights
- Ability to monitor remote PBXs from 3CX Assistant
- Ability to transfer calls from one 3CX PBX to another PBX using the Multi-PBX capability of 3CX Assistant
- Point and click creation of conference calls from 3CX Assistant
- Ability to directly leave voice mails for other users from 3CX Assistant
- Ability to require a PIN to create a conference call
- Ability to rewrite caller IDs per port
- Ability to call a users mobile number directly
- Ability to hide certain extensions from the 3CX Assistant view
- System wide black list
- Click 2 dial now calls phone in intercom mode, making call setup much faster
- Full CRM integration with Microsoft Dynamics
- Myphone portal has been improved
- Ability to customize the colors of 3CX Assistant
- Ability to define a status (Away, Out of Office, Available) with corresponding call forwarding rules
- Ability to specify more then one fax extension

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 Help pages

Additional documentation, tutorials and documents about 3CX Phone System can be found on our help pages/ knowledge base, located at:

http://www.3cx.com/blog/help

2. Installing 3CX Phone System for Windows

System requirements

3CX Phone System for Windows requires the following:

- Windows XP Pro, Vista Business, 7 , 2003 server, 2008 server or 2008R2
- Port 5060 (SIP), 5090 (Tunnel - Optional) to be opened
- Port 5480, 5481, 5482, 5483, 5484, 5485, 5486, 5487,5488 to be free
- One port for IIS default 80 or Abyss default 5481
- .NET Framework version 3.5 or higher
- 1 Gigabyte Memory or higher, Pentium 4 processor or up
- Internet explorer v7 & up, Firefox 2 & up
- If using a VoIP provider or remote extensions, you will have to a fixed IP. DynDNS configurations are not supported!

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 100 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

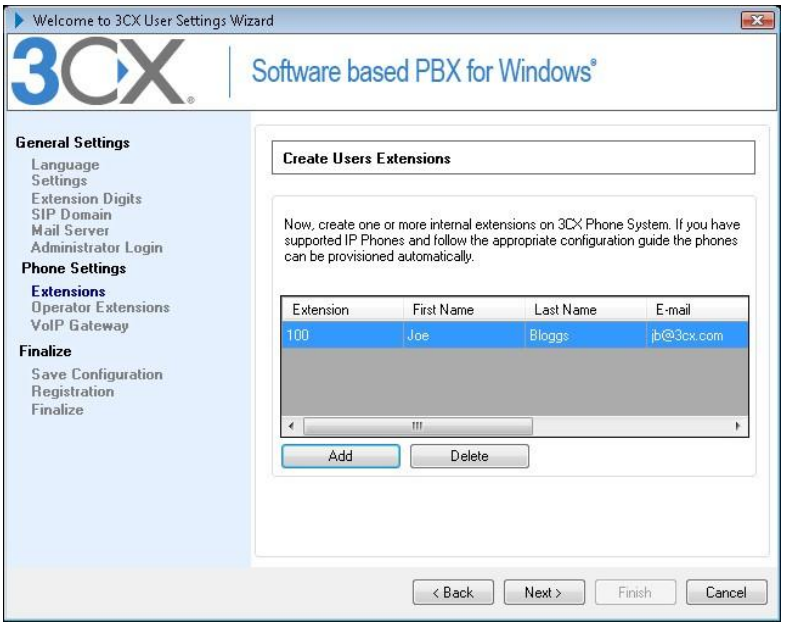
You will be asked whether you wish to use IIS or the in-built web server, Abyss. With Abyss your phone system will be independent of windows updates and other web sites

running on IIS. Abyss is equally performant. On windows XP only Abyss can be selected due to the limitations of IIS 5.1)

4. 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. 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.
4. If you have more then one network interface, set-up will ask you on which network interface 3CX should listen.
5. Set-up will ask you for the public IP of the 3CX Phone System machine so that remote extensions can be provisioned. You can skip this step if you wish and enter it later.



Screenshot 1 - The 3CX Configuration Wizard

6. 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

7. 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
8. Now specify the voice mail number – this is the number that users will call to retrieve their voice mail.
9. 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.
10. 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 Abyss, 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 8, 7.1 or 7.0 installation to version 9 as follows:

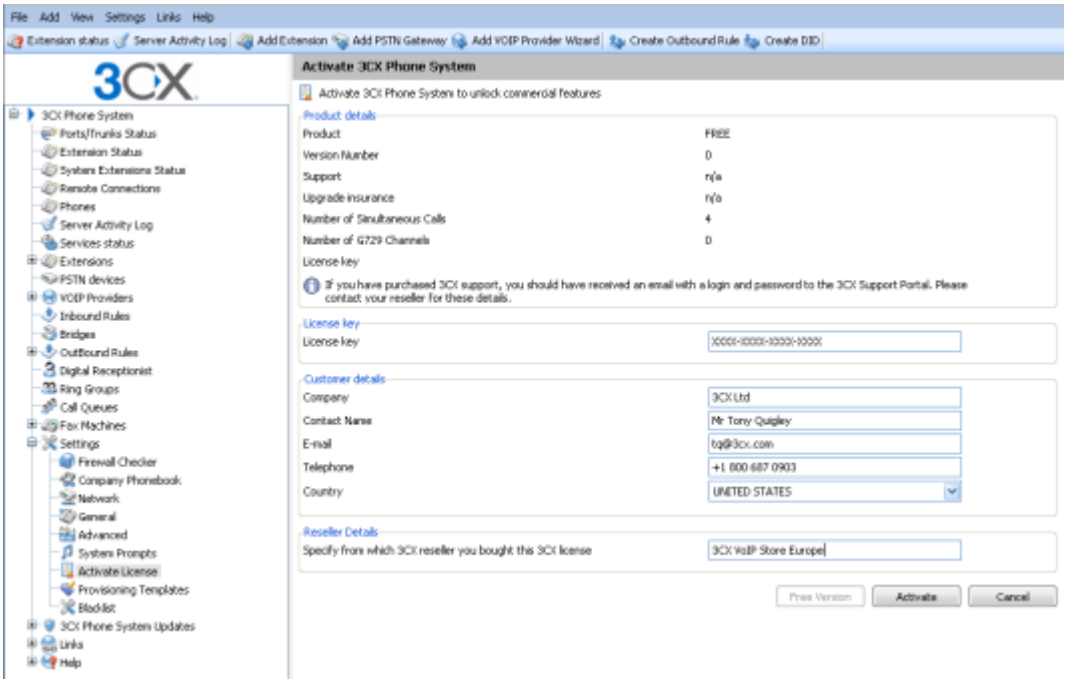
1. Before uninstalling, backup your current configuration using the backup and restore tool located in the 3CX program group.
2. Now uninstall using Add/Remove programs.
3. Install 3CX Phone System v9 – You will be prompted for the backup file by the configuration wizard. This will restore your configuration.

Notes in regards to upgrades from older versions:

- Upgrades of version 3.1, 5, 6.0, 6.1, configurations are not supported.
- 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.

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 Reseller name 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 on a new machine or when a change in the local network topology occurs (example: the local ip address changes)

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/blog/voip-howto/linksys-router-configuration/
- Port 5061 (TCP) for TLS communications – If using secure SiP.
- Port 5090 (UDP and TCP) for the 3CX Tunnel (if you plan to use remote extensions)
- Port 9000-9049 (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.

The 3CX Extension user manual

The 3CX Extension manual explains users how the users 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 extension manual is available online at: http://www.3cx.com/blog/um9en/

The 3CX Assistant

The 3CX Assistant is a compact utility which allows you to easily manage your extension with a few mouse clicks – rather then via a cryptic and limited phone interface. 3CX Assistant is not a phone – it works in tandem with a desk phone or soft phone. For more information about the 3CX assistant and how to install it, please refer to the chapter '3CX Assistant'.

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 3CX Phone System Updates > System prompt sets node and then select the prompt set you wish to use. The prompt set will be downloaded.

2. Now go to the Settings > System prompts sets 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.

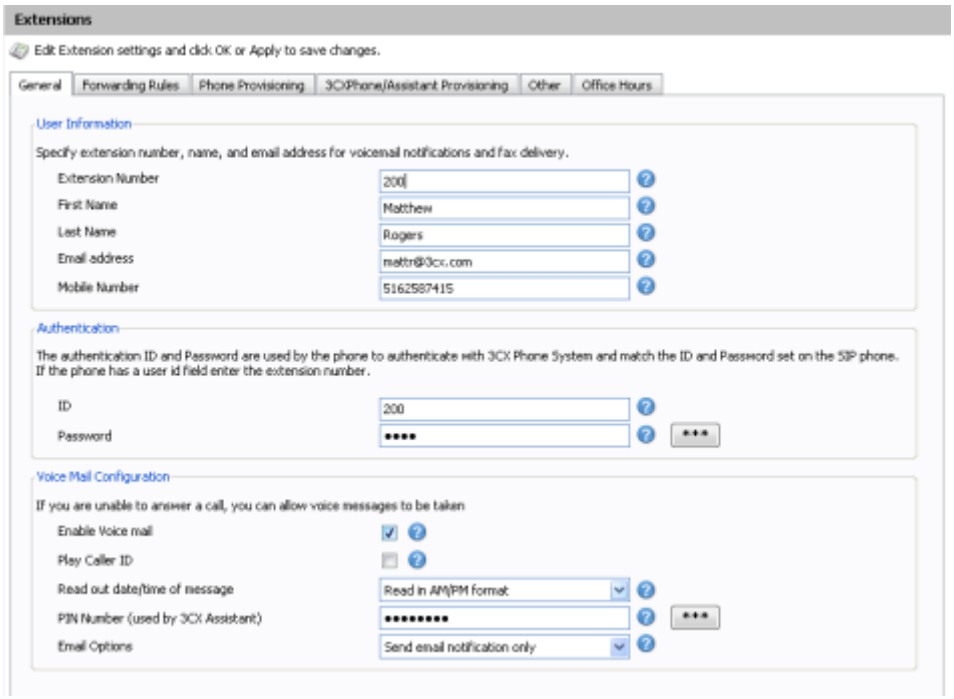
Start-up the 3CX Management console from the 3CX program group, or point your web browser to the Management Console by entering the name of the machine, for example: http://phone-system/Management.

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.



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.

Forwarding rules

Each extension can have a set of call forwarding rules that defines what 3CX Phone System should do when an extension user is unable to take an incoming call. This can be configured based on the user's status, the time, whether its an internal or external call or the caller ID.

Each status requires a call forwarding rule. For example, if the user is unable to take a call whilst his/her status is 'Available', you can forward the call to voice mail, whilst if the status is set to 'Out of Office' you could forward it to his/her mobile.

Call forwarding can be configured by the administrator from the management console or by the user from the MyPhone portal. Please see this web page for instructions how to configure call forwarding for an extension:

http://www.3cx.com/blog/um9en/myphone/forwarding-rules/

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.

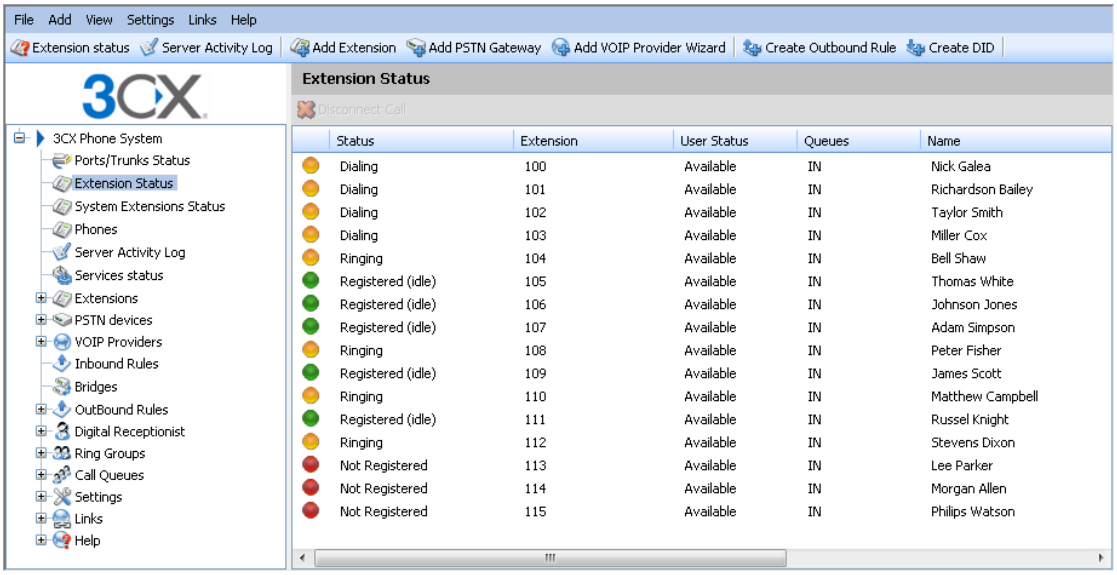
Importing extensions

If you need to create a large number of extensions it is handier to bulk import the extensions. To do this, you can create a spreadsheet with columns for each field that you wish to import. Alternatively you can import directly from Active Directory via LDAP. Please find a detailed description of fields as well as a link to a sample import file at: http://www.3cx.com/blog/docs/bulk-extension-import/

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



Screenshot 4 - The Status monitor

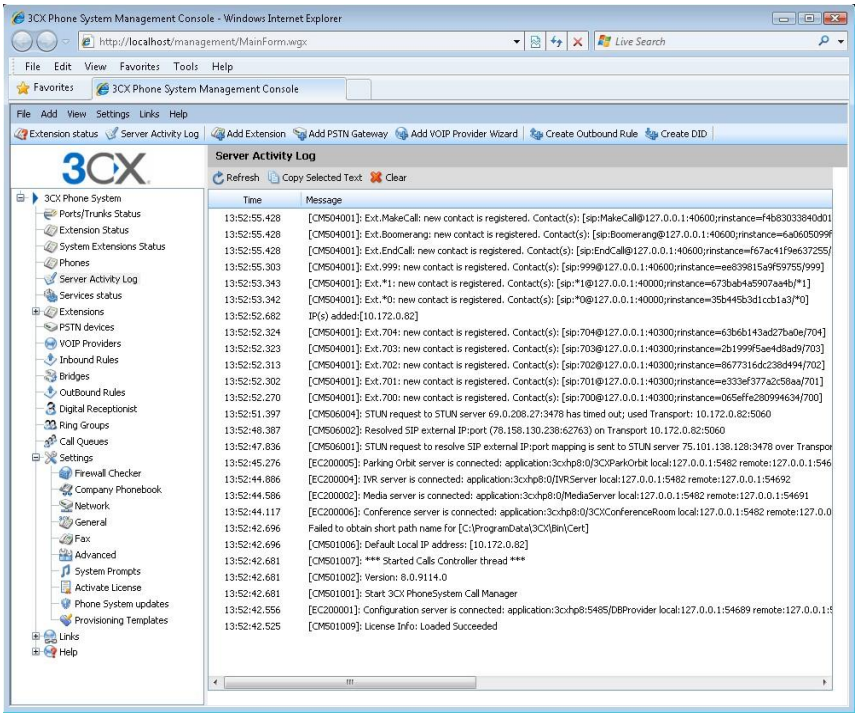
Load up the 3CX Phone System Management console, and click on the 'Extension Status' node. Check that all extensions are listed and are 'Registered'.

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.

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 easily manage the phone system and phones from the 3CX Management Console.

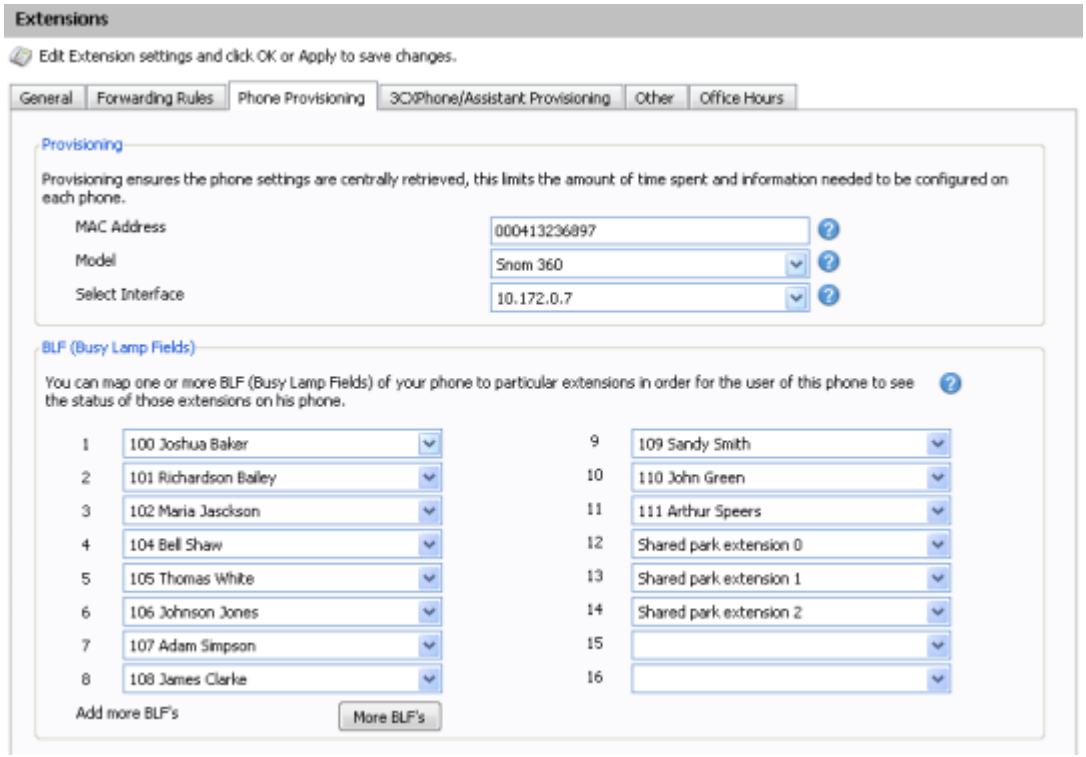
When using phone provisioning, it is easy to change extension passwords, BLF lights and so on because you can do it from the management console and then reboot the phone and push out the changes without having to look at the configuration of the phone itself.

Provisioning your phones is a two step process:

1. Create the extensions and specify the phone model and its MAC address. In the case of 3CXPhone and 3CX Assistant, the MAC address is not required.
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



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 configured. 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. You can also link a shared parking place to a BLF button. This allows users to easily park or un-park calls by hitting that BLF button.

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).

Configuring provisioning of 3CX Phone / 3CX Assistant

3CXPhone and the 3CX Assistant can be provisioned similarly to IP phones. To configure this:

1. Switch to the 3CXPhone / Assistant provisioning tab
2. Select the Network interface (in case your phone system server has multiple network interfaces)
3. Specify whether you wish 3CXPhone and 3CX Assistant to use the 3CX tunnel when they are being used remotely.
4. Specify whether you wish the 3CXPhone to be integrated with the 3CX Assistant
5. The provisioning files will automatically be created for each extension and the provisioning URL will be included in the welcome email sent to the extension user.

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 in 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 URL 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. You can find configuration guides for each phone here: http://www.3cx.com/sip-phones/index.html:

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 URL 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: 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.

Provisioning URL

The provisioning URL differs depends on which port your web server is running:

http://<ip of 3CX>:<port>/management/provisioning

e.g. http://10.172.0.2/management/provisioning if using port 80 or
http://10.172.0.2:5481/management/provisioning for port 5481

Each phone requires this URL to be specified in a particular manner. Please see this link for the exact format for each phone:

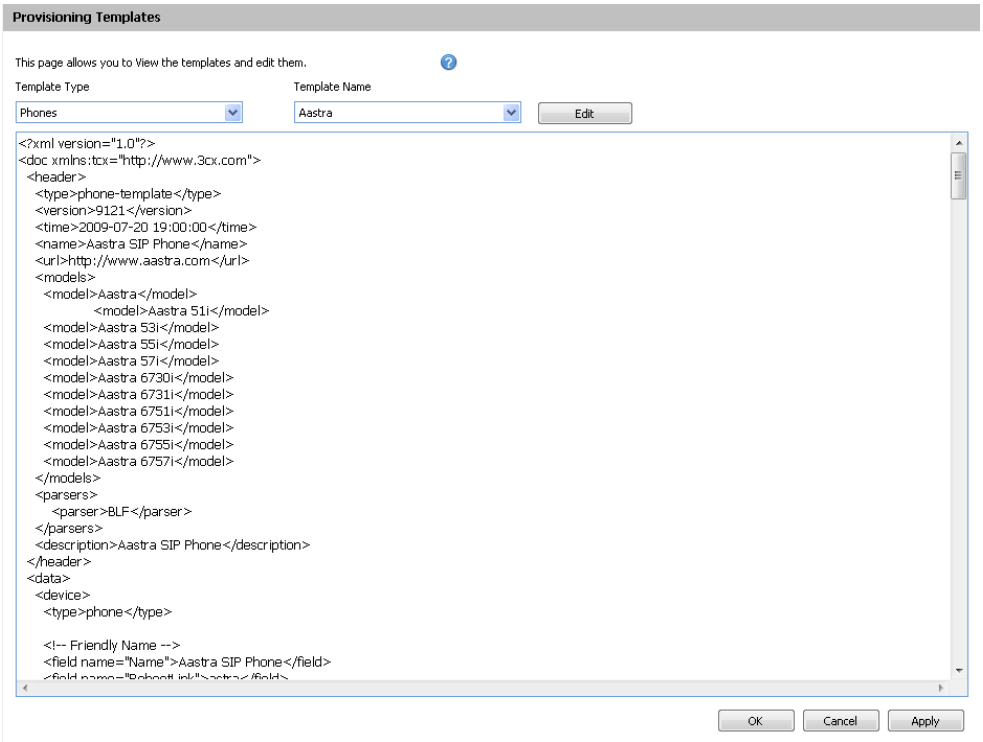
http://www.3cx.com/sip-phones/DHCP-option-66.html

Provisioning 3CX Phone / 3CX Assistant

To provision 3CXPhone and 3CX Assistant, all you need to do is specify the provisioning URL the first time you start 3CX Assistant. The provisioning URL is included in the welcome e-mail sent to the extension user. For detailed instructions follow this link

http://www.3cx.com/blog/um9en/3cx-assistant/installation/

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:

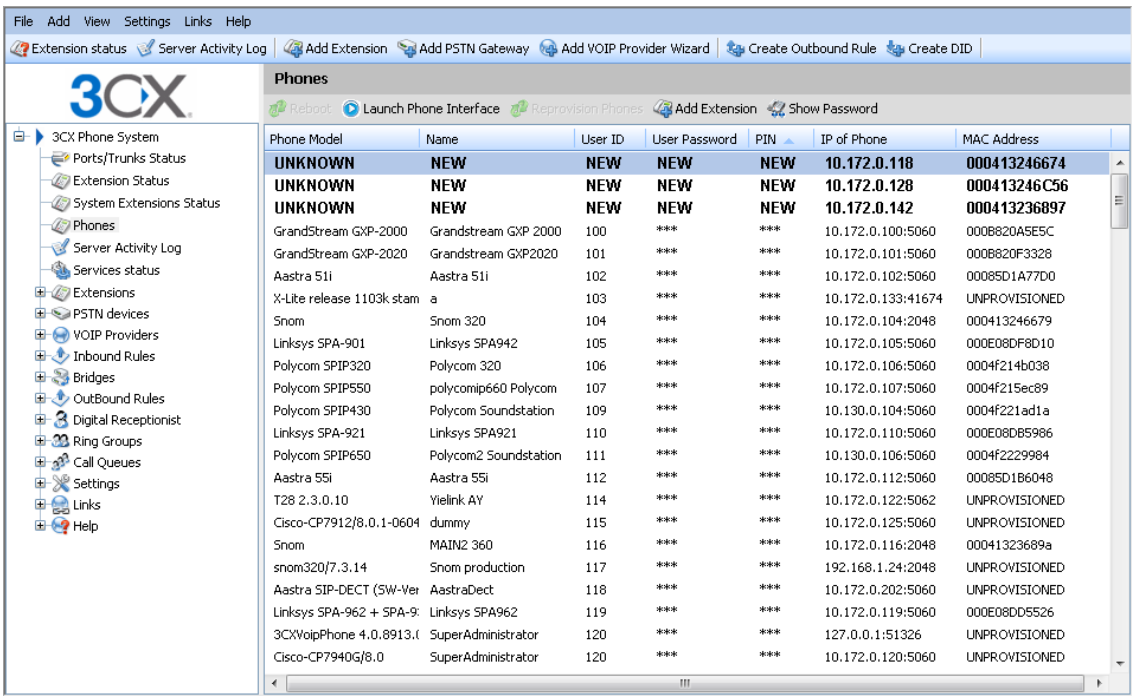
1. Go to the Settings > Provisioning templates node
2. In Template Type select ‘Phones’
3. In ‘Template Name’ select phone model
4. Edit the template as necessary and click OK to save the changes. To make the changes active, you need to re-provision the phones from the phones node.

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’
4. Select the phones again and select ‘Reboot’ to make the new provisioning information active on the phone.

Monitoring your IP phones



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:

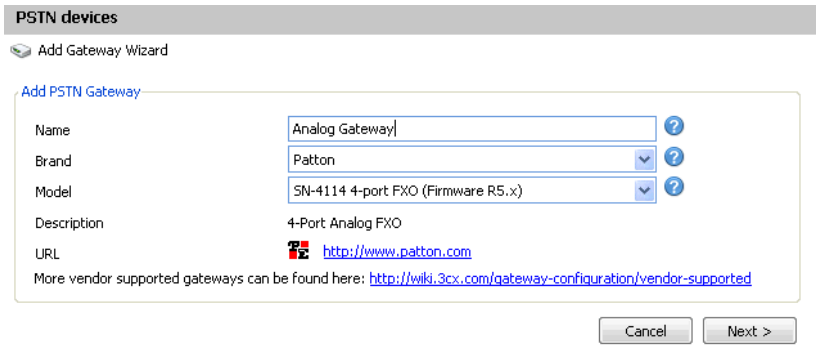
Port – a port is a physical port on a gateway or VoIP card. In the case of an analog line, one port equals one voice channel. In the case of BRI ISDN, one port is 2 voice channels, and in the case of E1 or T1 ports, each port represents 30 and 23 channels respectively.

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'

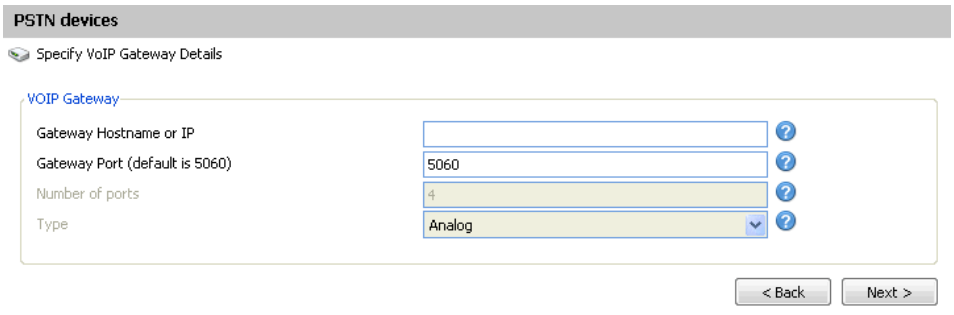
Step 1: Create the PSTN gateways in 3CX Phone System

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



Screenshot 9 – Choosing gateway template

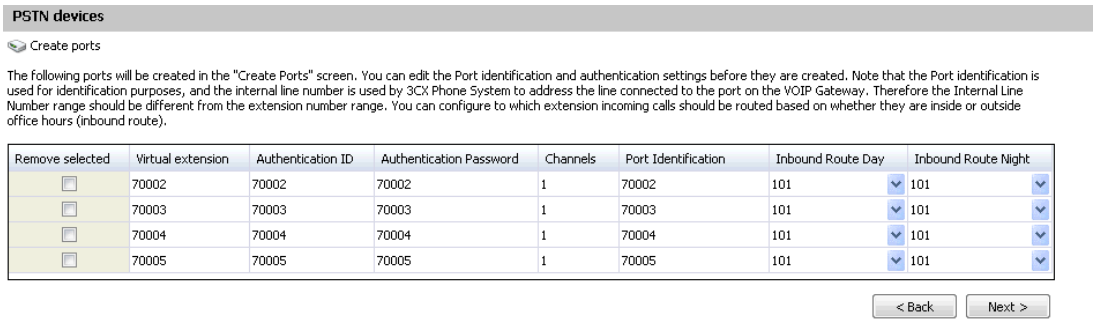
1. In the 3CX Phone Management console menu click on Add > PSTN Gateway.
2. In the name field, enter a friendly name for the VOIP gateway. Now choose the gateway brand and model that you are using from the list. If it is not listed, select generic. Note that we will not be able to support you in using 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.



Screenshot 10 - Specifying VoIP Gateway details

4. 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.

5. 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.



Screenshot 11 - Adding PSTN Lines

6. 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.

6. 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.

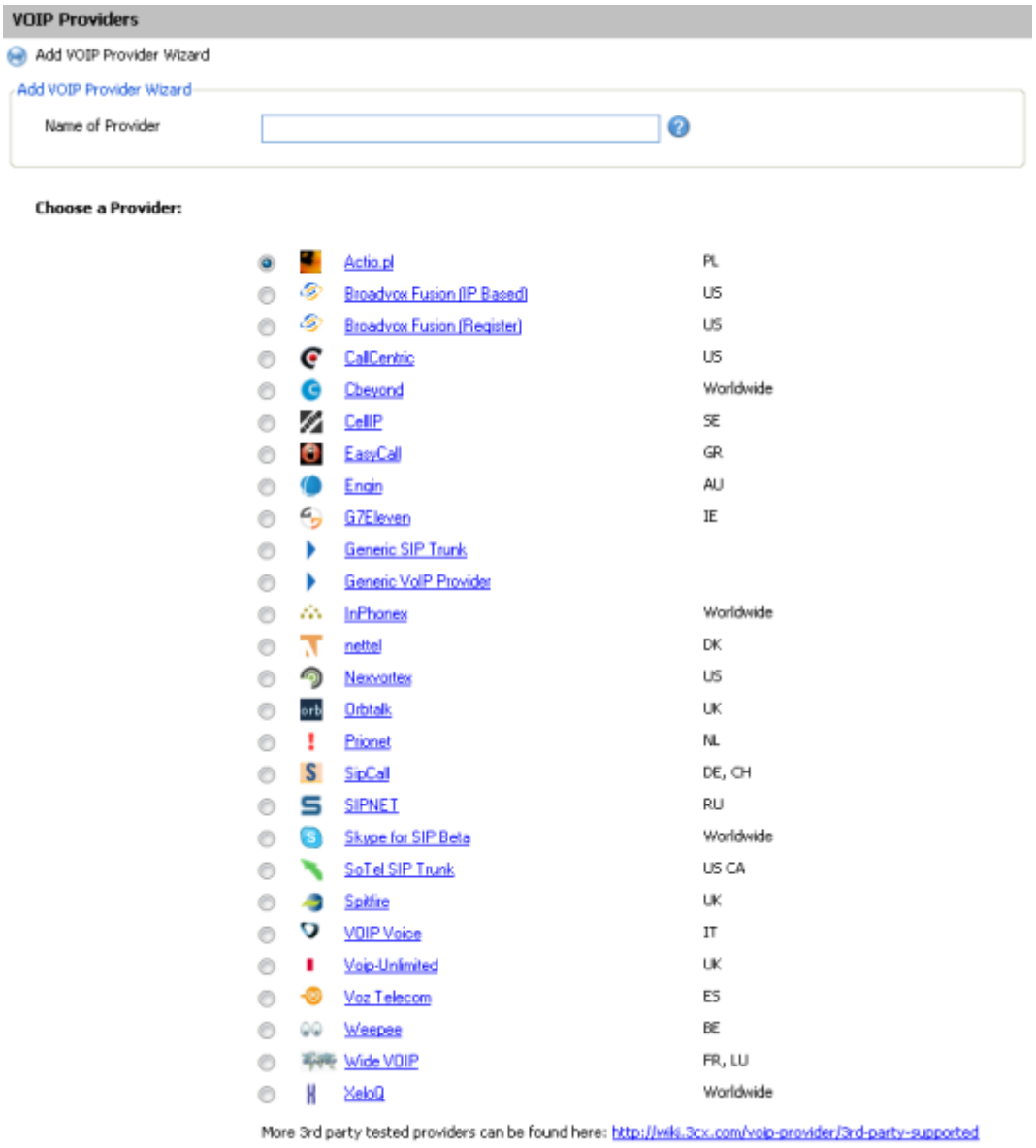
Note that you can use any SIP based provider but we will not be able to support you in the configuration of the VOIP provider, nor can we guarantee that it will work. Therefore, unless for your country no VOIP provider is listed, use a supported VOIP provider.

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 able to guarantee that it will work with this VOIP provider. Click Next.



Screenshot 12 - Add VOIP Provider account

3. 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.

4. 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.

5. Now specify how calls from this VOIP provider should be routed. You can specify a different route outside office hours.
6. 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.

Screenshot 13 - 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.

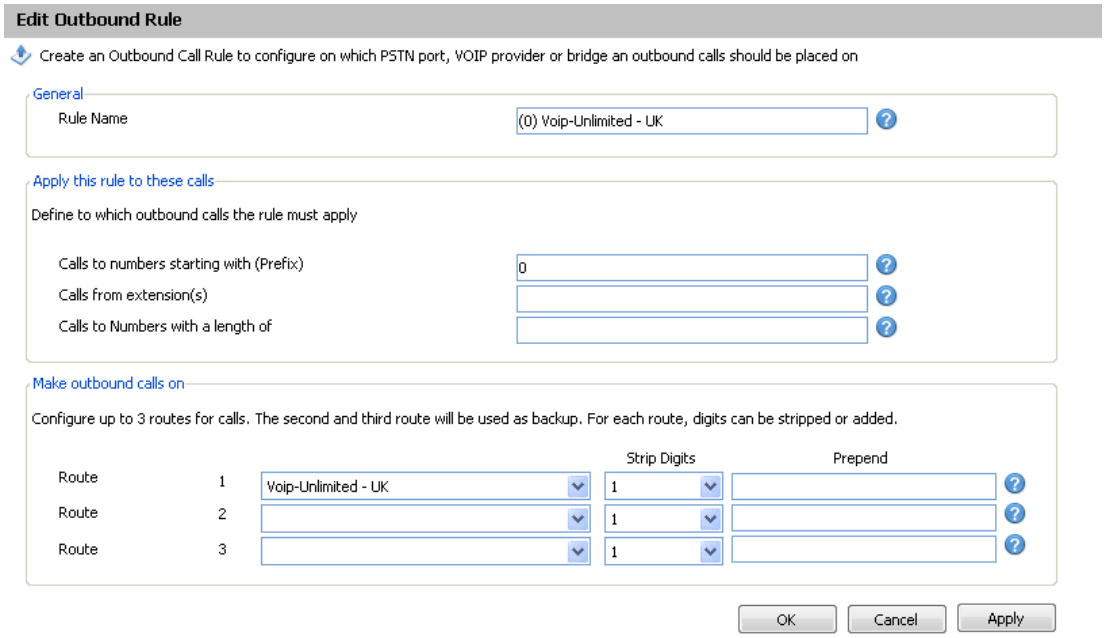
Note: Configuration is necessary because source identification based on DID and routing based on DID are two different things and thus require separate configuration.

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



Screenshot 14 - 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.

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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 then 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.

Deploying 3CX Assistant

The 3CX Assistant set-up needs to be run on the local machine. The latest set-up can be downloaded here:

http://www.3cx.com/downloads/3CXAssistantSetup9.exe

After set-up, 3CX Assistant will ask for the provisioning URL, so that 3CX Phone System can automatically configure the settings. This also allows you to easily change server settings, tunnel configuration as well as PIN codes or passwords.

The provisioning URL is created automatically and included in the welcome e-mail sent to the user when you create an extension. You can resend this email by going to the Extensions node in the management console, selecting the extension and clicking on the button 'Send Welcome Email'.

The installation instructions of 3CX Assistant can be found here:

http://www.3cx.com/blog/um9en/3cx-assistant/installation/

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. Go the 3CX Phone System Updates node in the management console

2. Click on 'Other updates'

3. Download any updates to the 3CX Assistant from the 3CX website. This will make the update available to all 3CX Assistants in the network.

4`. When the 3CX Assistant clients login, they will check for an update and if available automatically download it and install it.

Note: To update the 3CX Assistant immediately, select Help, Check for updates.

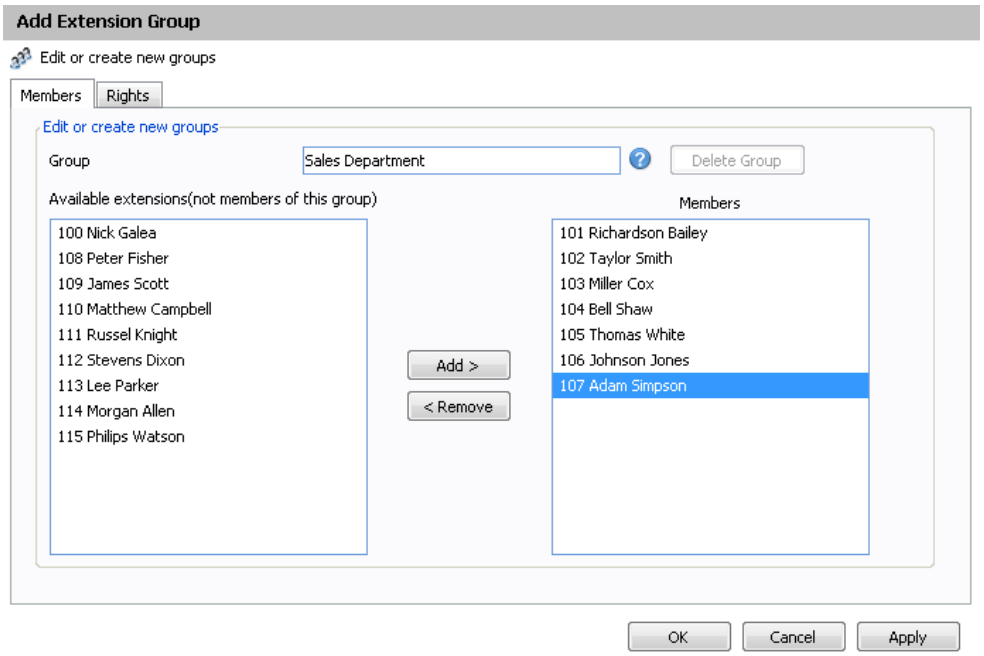
Using 3CX Assistant

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

http://www.3cx.com/blog/um9en/

Grouping & Assigning Rights

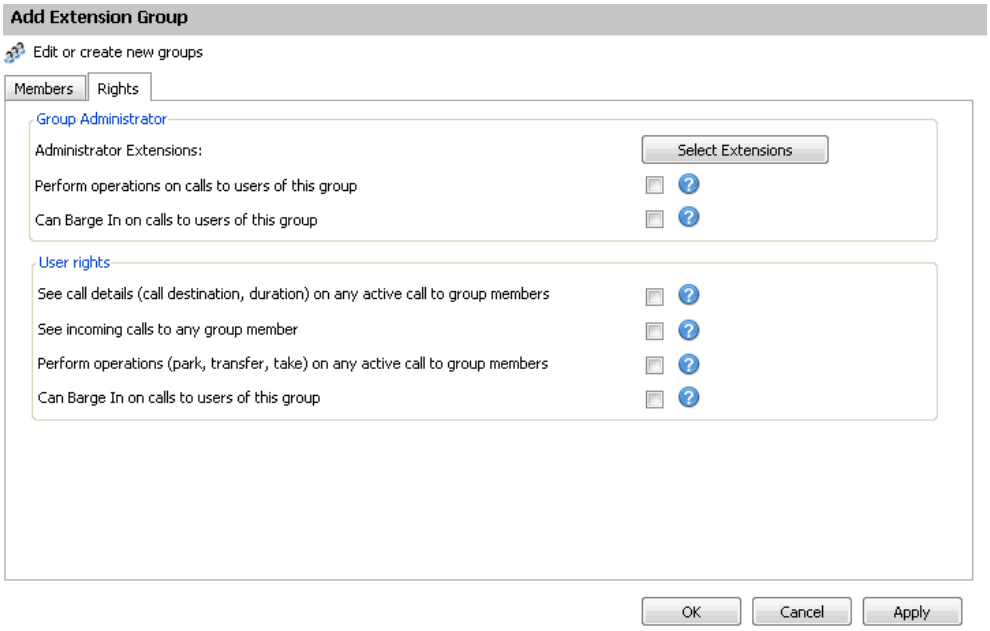
3CX Assistant can limit what call information is displayed based on extension groups. These extension groups are used to determine what information is shown to whom. In addition they help group the extension for both users and the administrator. To create an extension group:



Screenshot 15 - 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.



Screenshot 16 - 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.

Configuring additional connections

It is possible to connect to additional 3CX Phone Systems located in a remote office. You will need to have a separate account on the remote PBX. To do this:

1. Select ‘Connections’ from the file menu.
2. Now click ‘Add’. The Connection Details dialog will open up.
3. You can now add the provisioning URL of the extension on the remote PBX, or else configure the connection manually.
4. After you click OK, the 3CX Assistant interface will change to a tabbed interface, each tab representing a PBX.
5. You can switch between a PBX by click on a tab. If you wish to transfer calls to another PBX, drag the call onto the tab, after which the extensions in that PBX will show. You can then continue to drag the call on to an extension to transfer the call.

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 then “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 17 - 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.

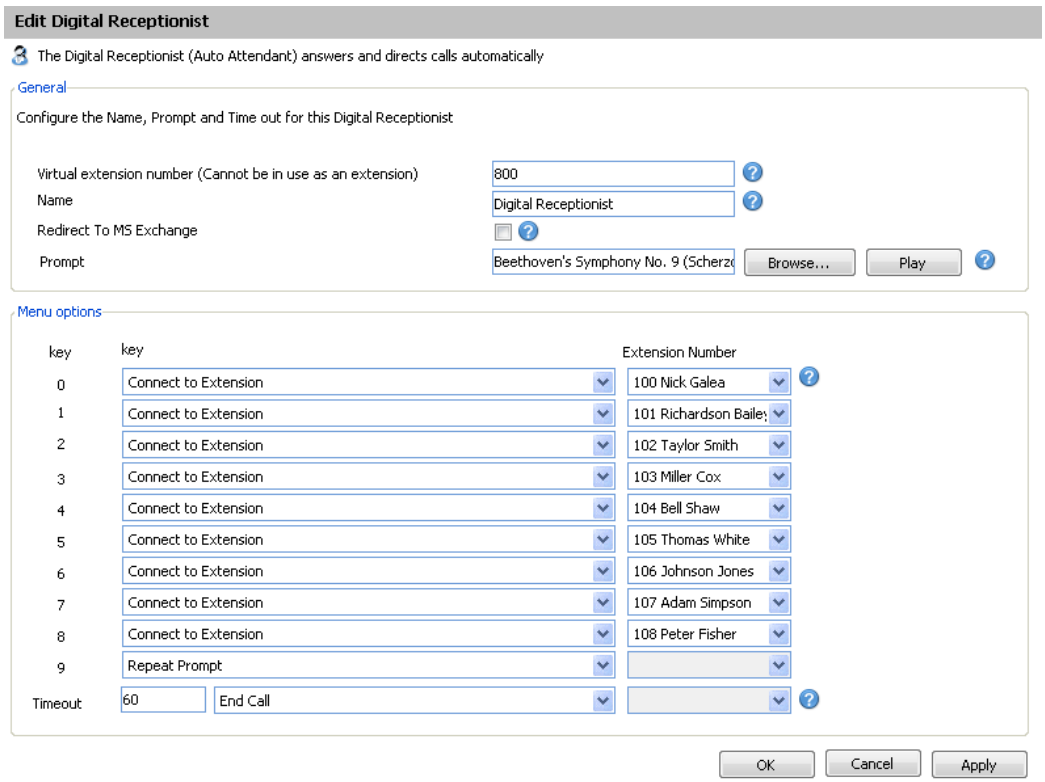
4. When creating a digital receptionist menu you will be asked for the path to the file.

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.



Screenshot 18 - 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\Ivr\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 are 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".

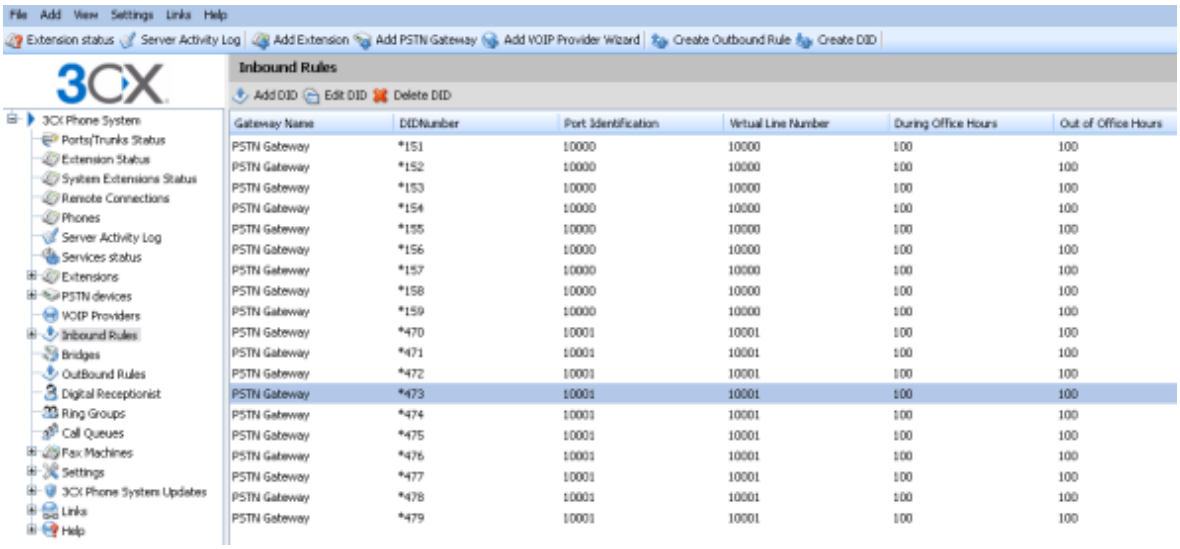
If the user does not input any more digits (2 seconds elapsed or '#' has been pressed) and more then one user is matched, then the IVR will play: "To call to Van Damme press 0. To call to Van Halen press 1. To exit press pound". In this example 'Van Damme' and 'Van Hallen' are self-identification prompts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

10. 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.

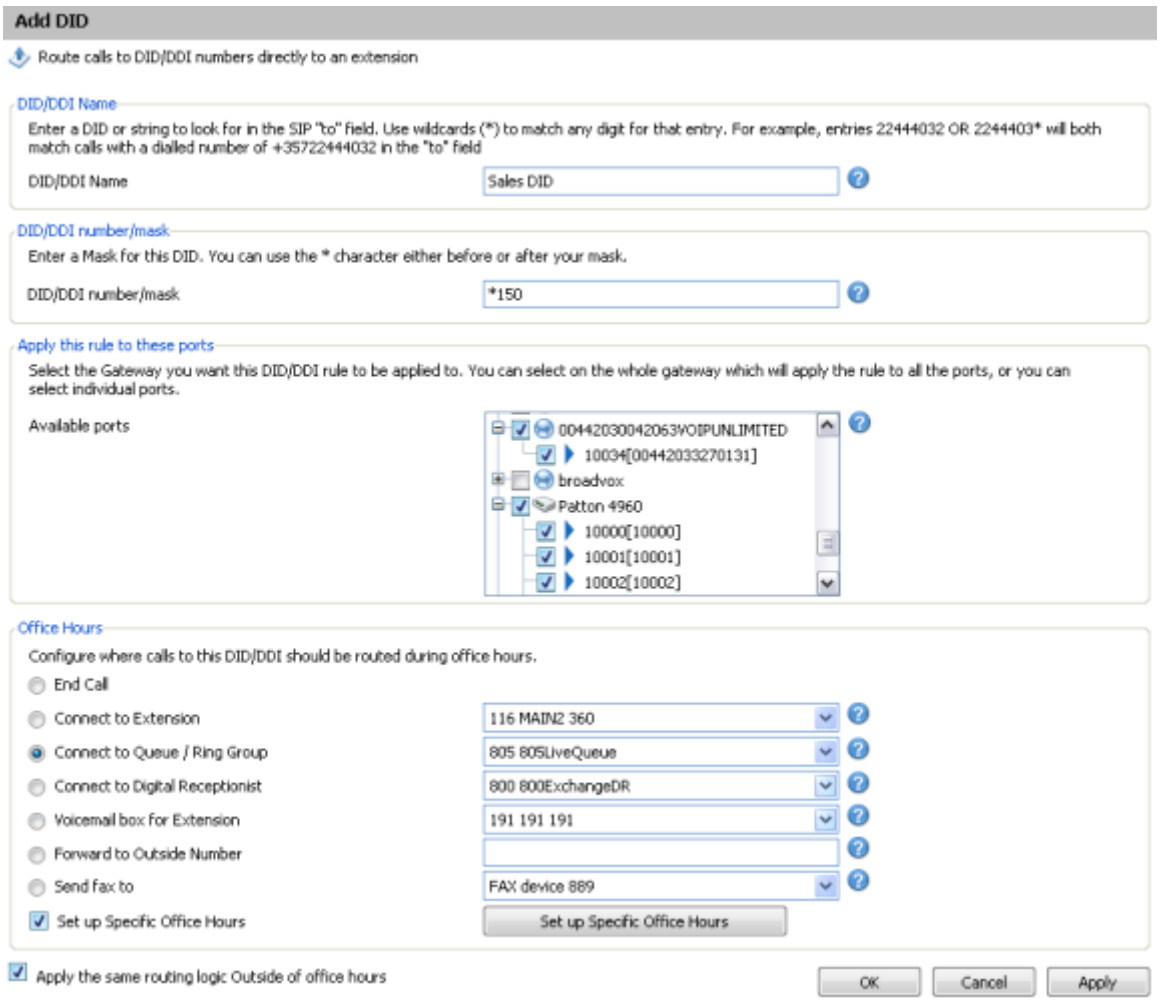


Screenshot 19 - Configuring DID’s

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



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

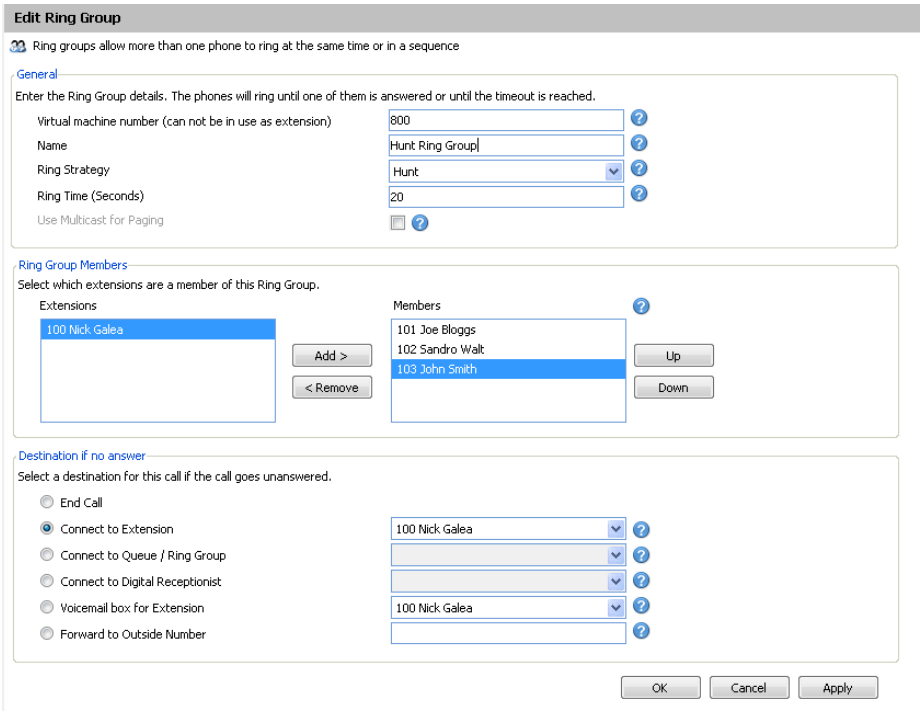
DID number into the “To” field. In the case of a VOIP Provider, contact your VOIP provider for more information.

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.

11. 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.



Screenshot 21 - 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.

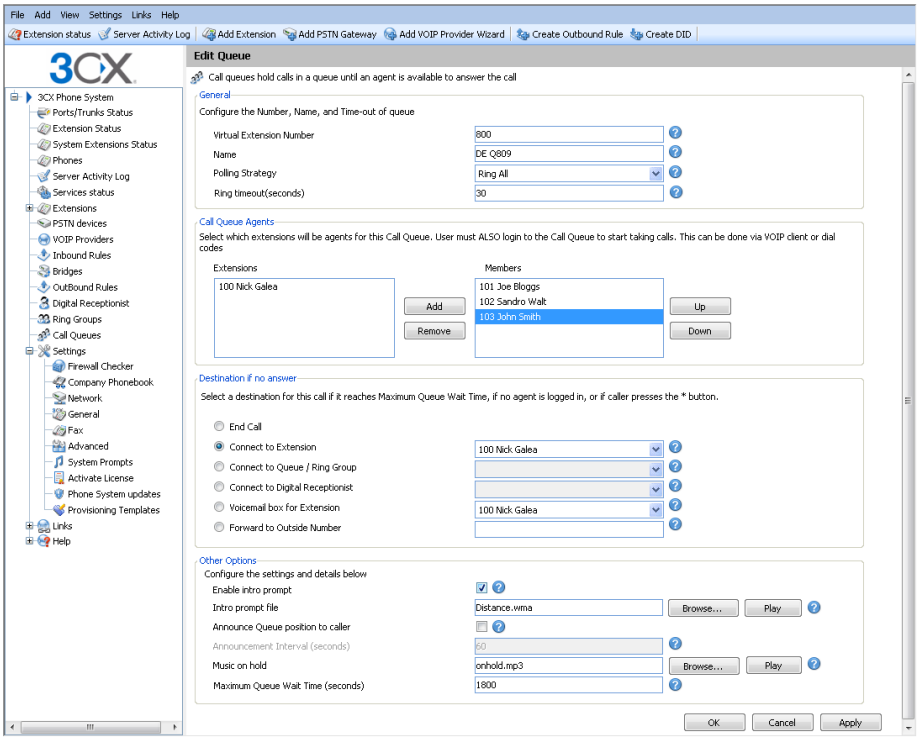
To call a user via the intercom function:

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 22 - 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.

12. 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.

Configuring conferencing

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.
4. Now specify whether you wish to require a PIN to create a conference. If you enable this, users that **create** a conference must enter this conference PIN after the conference ID when creating a conference.

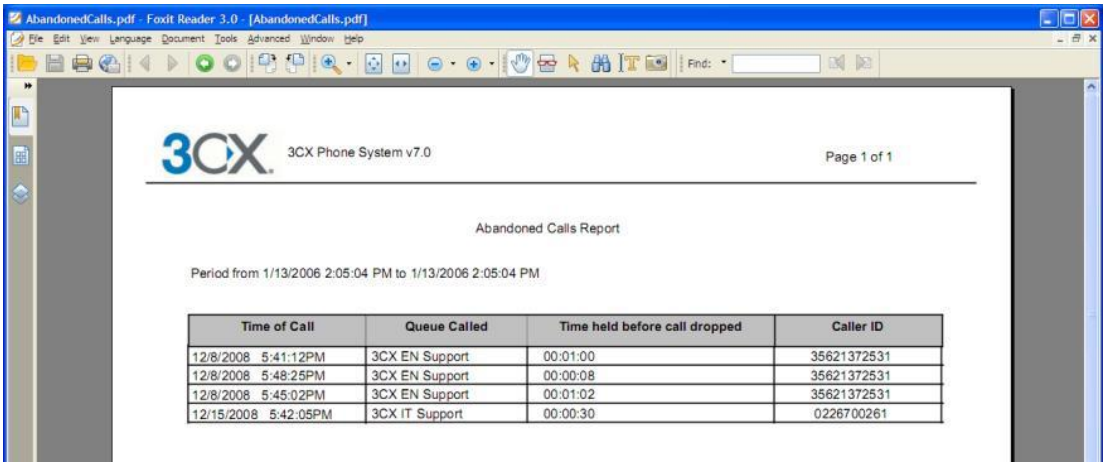
Creating a call conference

For information how to create a call conference, see this page of the online extension user manual:

http://www.3cx.com/blog/um9en/conference-calls/

13. Generating Call Reports

Introduction



Screenshot 23 - 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

14. 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

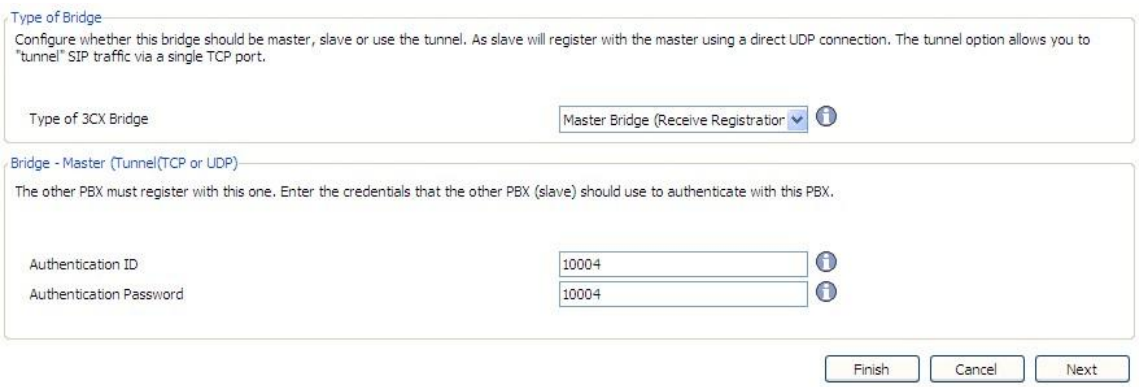
To create a bridge:

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.

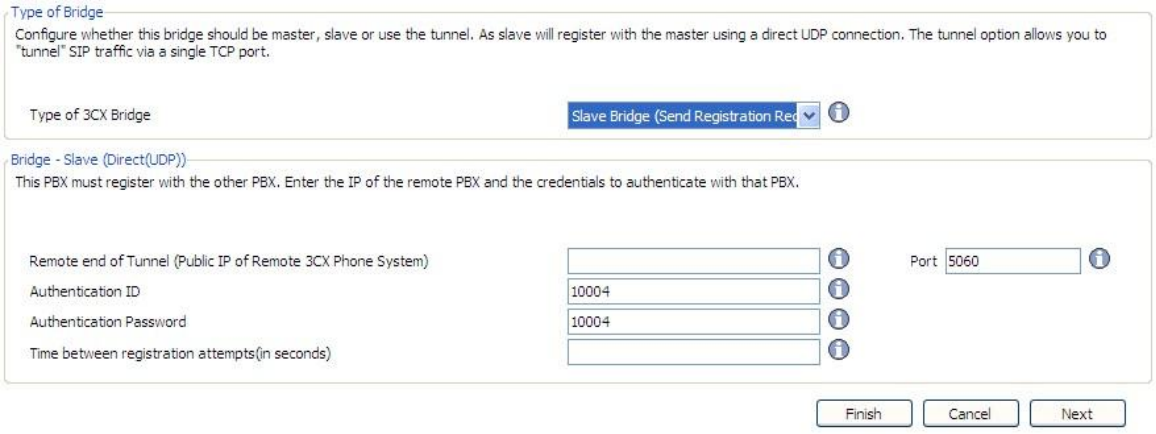


Screenshot 24 - Creating a master bridge

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.

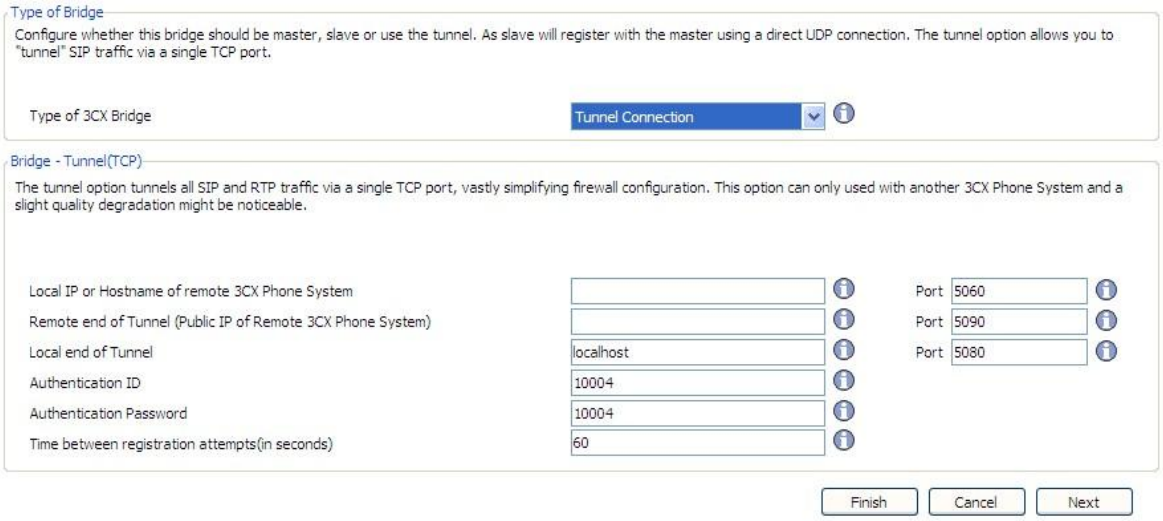


Screenshot 25 - Configuring a slave bridge

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.



Screenshot 26 - 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 much 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.

15. 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.

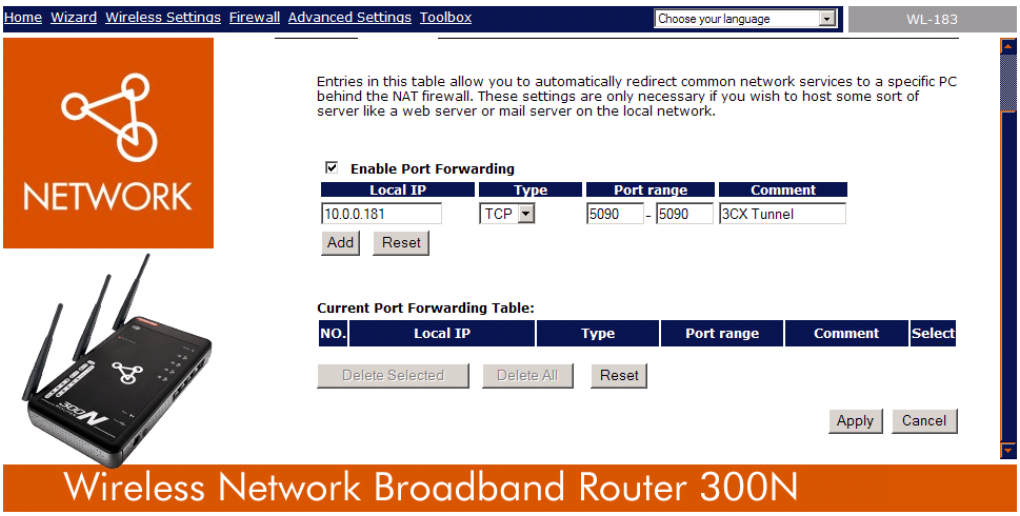
Step 1 – Configure the PBX

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.



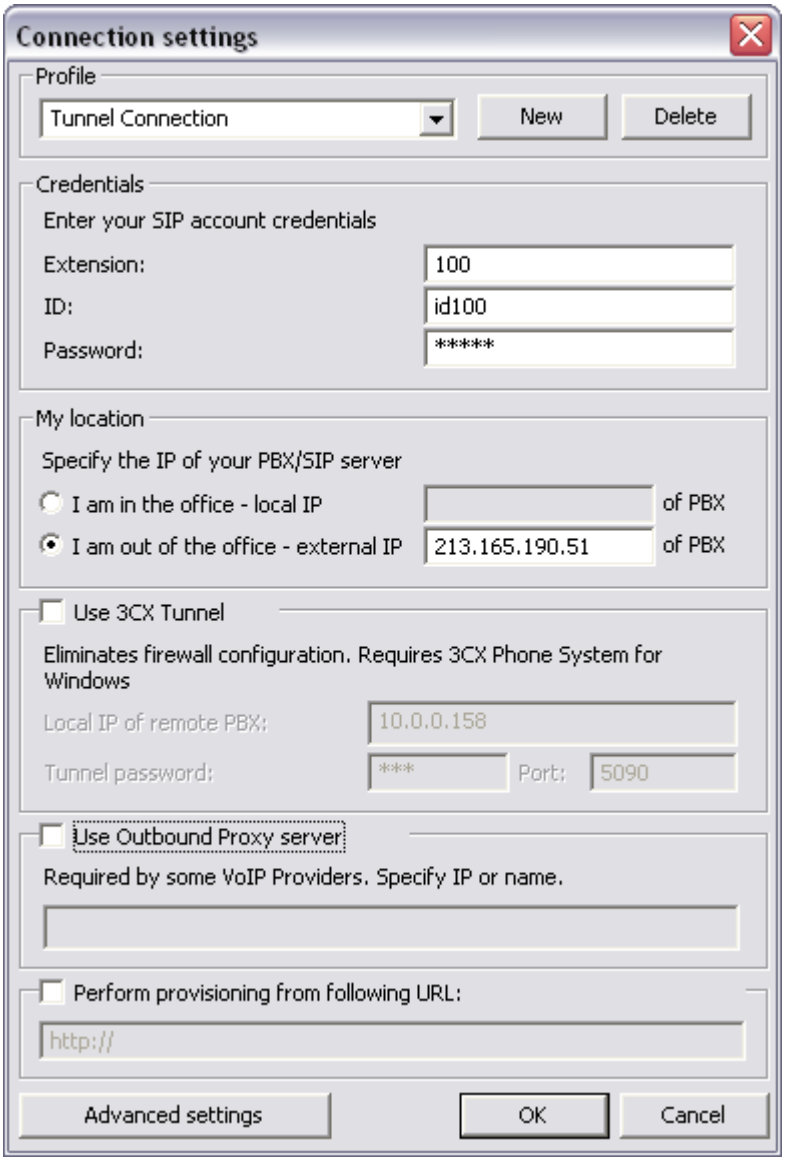
Screenshot 27 - Configure a port forward rule

The above picture shows configuration for a Sitecom WL-183 WAN-to-LAN router, but most routers will provider 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



Screenshot 28 - 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:

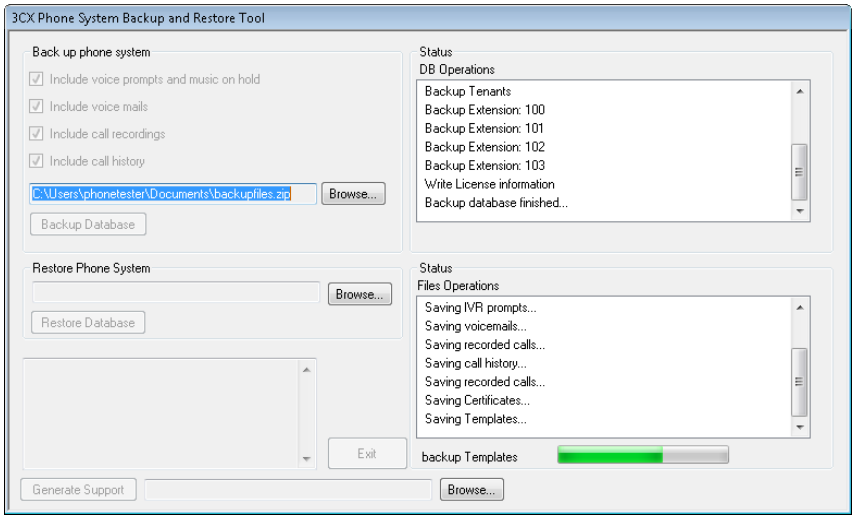
http://wiki.3cx.com/documentation/networking/sip-proxy-manager

16. 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 29 - 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

17. Fax server

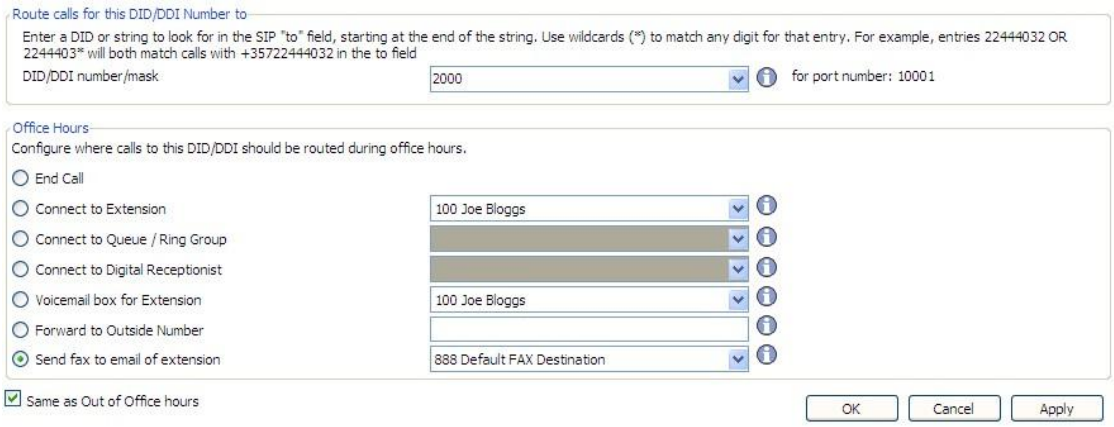
Introduction

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

3CX Phone System includes a fax server that allows receiving of faxes. The 3CX fax server is based on the T38 standard and requires a compatible supported T38 VOIP gateway or provider. Note that 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

To receive faxes, 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.



Screenshot 30 - 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.

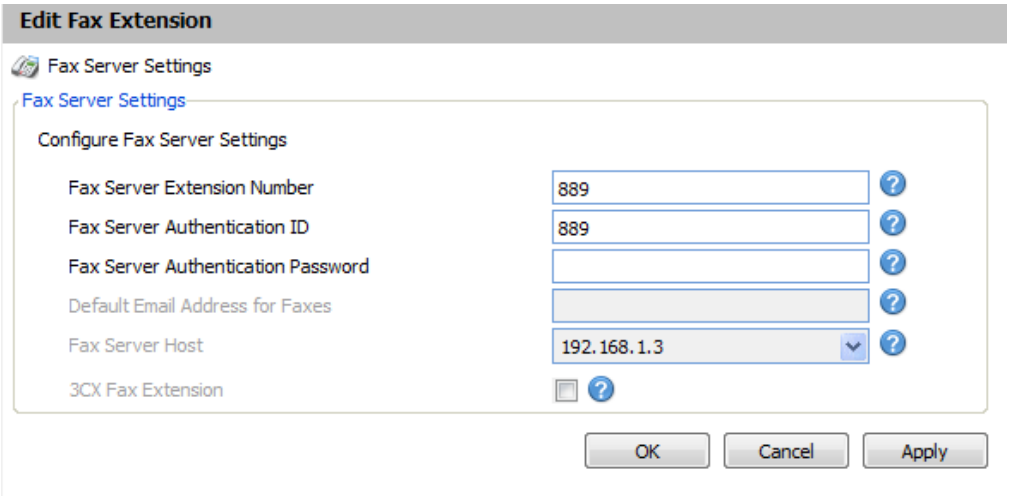
2. Select ‘Send fax to email of extension’

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.

Fax machines node

The fax machines node in the management console lists all known 'Fax extensions' including the extension used by the 3CX Fax Server. These 'Fax extensions' are similar to a normal extension and require an authentication ID and password to login to the SIP server. In addition, 3CX Phone System proxies T38 traffic to them. You can create additional fax extensions for fax machines connected to an ATA or for software based T38 fax servers.

Fax extension settings



Screenshot 31 - Fax extension

To edit fax extension settings or create a new fax extension:

1. In the 3CX management console, go to the Fax machines node. Now edit the existing fax extension (default 888) or add a new one.
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.
4. If this extension is used by 3CX Fax Server, enable the option '3CX Fax Server'. You can specify the default email address to which all faxes should be sent.
5. Make sure the network interface that 3CX must use is selected
6. If this extension is used for an ATA or 3rd party T38 fax software, just un-tick the box. In this case the extension will be used only to register to the SIP server and receive T38 fax traffic.

Note: You must restart the fax service for changes to take effect.

18. The Phonebook

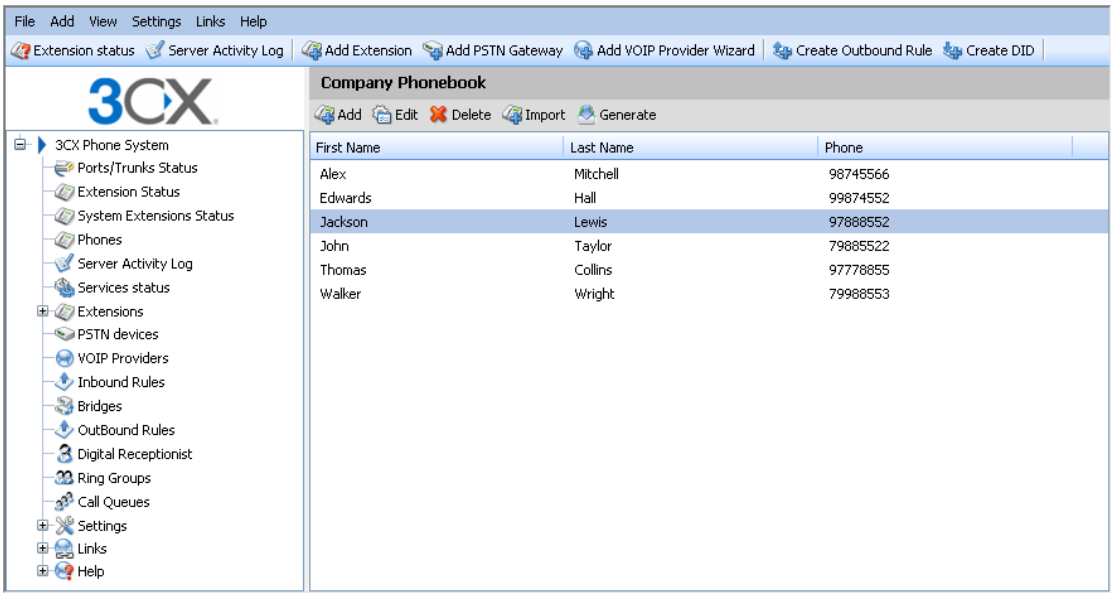
Introduction

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



Screenshot 32 - 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.

19. 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 33 - 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

20. Troubleshooting & Support

Introduction

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

- The 3CX KB / Help - http://www.3cx.com/blog/help/
- 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)

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

Knowledge base / Help page

3CX maintains a knowledge base / help page on its blog at

http://www.3cx.com/blog/help/

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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