Manual

3CX Phone System for Windows

Version 3.1

http://www.3cx.com

E-mail: info@3cx.com

Version 3.1 – Last updated 15 June 2007

Table of Contents

1. Introduction to 3CX Phone System for Windows 9
   - What is 3CX Phone System for Windows? 9
   - Benefits of an IP Phone System / IP PBX 9
   - How an IP Phone system works 11
   - SIP phones 11
     - Software based SIP phones 12
     - Hardware SIP Phones 12
     - Analogue phones using an ATA adapter 13
   - 3CX Phone System components 13
   - 3CX Phone System editions 13
2. Installing 3CX Phone System for Windows 15
   - System requirements 15
   - Run set-up 15
   - Activating 3CX Phone System 16
   - Firewall configuration 16
     - 3CX Phone System behind a NAT / firewall 17
     - Firewall installed on the 3CX Phone System machine 17
   - Installing the 3CX Call Assistant 18
   - The 3CX Call Assistant User manual 18
3. Creating Extensions 19
   - Introduction 19
   - Adding Extensions 19
   - Configuring the SIP phones 23
   - Testing your setup 23
     - 1. Check Phone System status with the status monitor 24
     - 2. Place a call to another extension 25
     - 3. Troubleshooting tips 25
   - Importing extensions 25
4. Adding PSTN line(s) via a VOIP Gateway 27
   - Introduction 27
     - What is a VoIP Gateway? 27

VOIP Gateway configuration overview ... 27
Step1: Create the PSTN lines in the 3CX Phone System ... 27
Step 2: Configure the VoIP gateway ... 30
Generating a configuration file ... 30
Configuring the PSTN Interface ... 30
5. Adding lines hosted by a VOIP Provider. ... 31
Introduction ... 31
Step 1: Create an account with a VOIP Provider ... 31
Step 2: Add the VoIP provider account in 3CX Phone System ... 31
6. Creating Outbound Call Rules ... 33
Introduction ... 33
Creating an outbound call rule ... 33
Manage outbound call rules ... 34
Examples of Outbound Rules ... 34
Routing calls by destination prefix example ... 34
Routing calls by dedicated line prefix example. ... 35
7. The Digital Receptionist / Auto attendant ... 37
Introduction ... 37
Recording a menu prompt ... 37
Recording the prompt in Windows Sound Recorder ... 37
Creating a prompt using Text to Speech ... 38
Creating a digital receptionist ... 38
Allowing callers to dial a known extension directly ... 39
8. Adding Direct numbers using DID/DDI lines ... 41
Introduction ... 41
DID numbers and 3CX Phone System – How it works ... 41
Adding a DID line ... 41
Troubleshooting DID lines ... 43
9. Ring groups & Call Queues ... 45
Ring Groups ... 45
Call Queues (Enterprise edition only) ... 46
10. Generating Call Reports ... 49
Introduction to reports ... 49

Creating a report 49
Analyzing the call logs 50
11. General configuration 53
Introduction 53
Admin credentials 53
General Settings 53
SIP Port 53
Ports to use for internal calls 53
Ports to use for external calls 53
Voicemail settings 53
STUN Server Options 54
Settings for direct SIP calls 54
Logging Level 54
Configuring office hours 54
Other 55
Exchange 2007 Unified Messaging Integration 55
System prompts 56
Backup and Restore 56
12. Troubleshooting 57
Introduction 57
Manual 57
Configuration guides 57
FAQ 57
Web Forum 57
Request support via our support system 58
Index 59

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.

Companies are switching their traditional phone systems / PBXs to IP PBXs at a staggering rate: IP Telephony equipment sales are increasing each year by more then 50% and are expected to reach $15 billion yearly by end 2007.

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 computers 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.

Scaleable:

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

Better customer service & productivity:

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

Software based Phones are easier to use:

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

More features included as standard:

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

Better control via better reporting:

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

Better overview of current system status and calls:

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

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

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

Allows easy roaming of users:

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

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

How an IP Phone system works

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

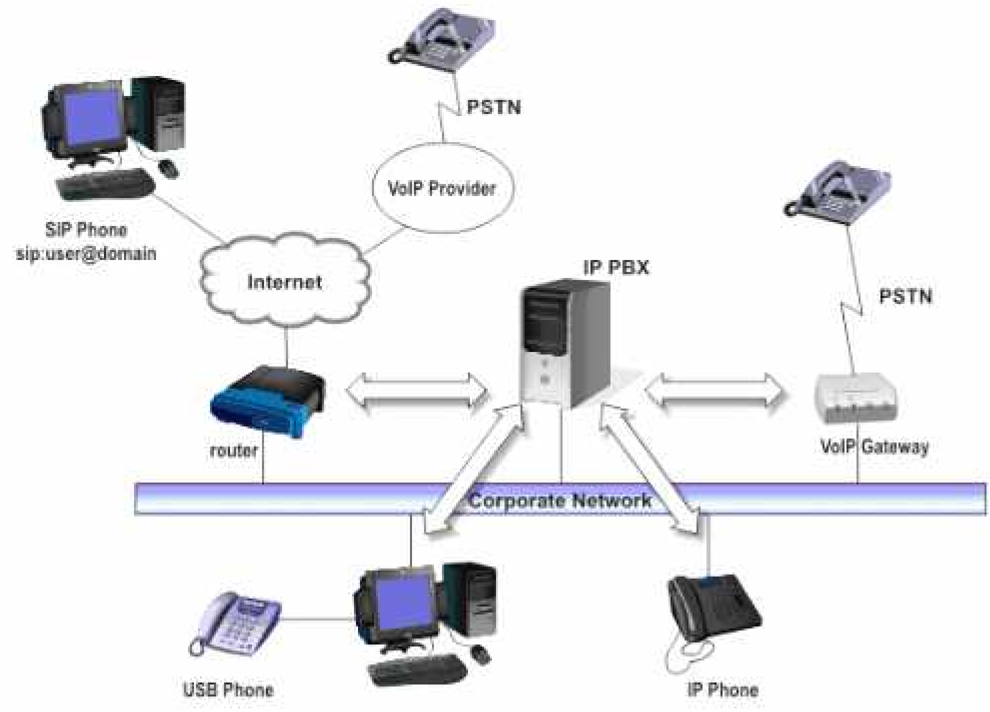


Figure 1 - VOIP Phone System Overview

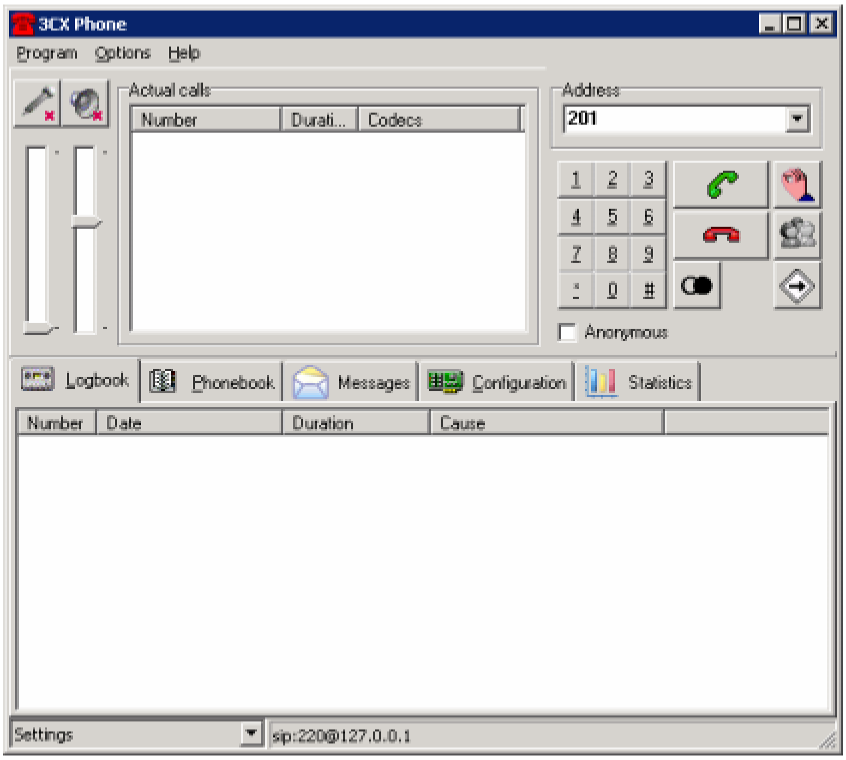
The image illustrates how an IP PBX integrates on the network and how it uses the PSTN or Internet to connect calls.

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

SIP phones

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

Software based SIP phones



Screenshot 1 – A SIP software phone

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 SJPhone from SJlabs, X-Lite from Counterpath, or 3CX VoIP Phone for Windows.

Hardware 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 CISCO 7940.

Analogue phones using an ATA adapter

Figure 3 - An ATA adapter

If you want to use your current phone 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 components

3CX Phone System consists of the following components:

- The SIP server service – this is a Windows service that sets up the calls using the SIP protocol. It performs the PBX functions such as call routing, call transfer and so on.
- A Media server service – this is a Windows service that performs call streaming, i.e. the actual audio conversation.
- A Management Console – to allow for web based configuration of the phone system. 3CX Phone System includes Apache web server, which is fast, scalable and secure.
- A database server service (Postgresql) – this is a light-weight SQL database server that stores all the phone system configuration settings.
- The Digital receptionist service – This services can answer calls and offer callers options.
- The Voice mail manager service – This service manages the voice mail boxes.
- The 3CX Call Assistant – This is a light weight windows client which allows users to manage their extensions and calls from their desktop.

3CX Phone System editions

3CX Phone System is available in 4 different versions – a Free 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

2. Installing 3CX Phone System for Windows

System requirements

3CX Phone System for Windows requires the following:

- Windows XP, Vista, 2000 (server & professional) or 2003 server
- Ports 5060 (SIP), 5481 (Apache) to be free and open and 5480 (Postgres), 5482 (Media server) to be free
- SIP standard based software or hardware phones
- Optional VOIP Gateway (if you need to connect PSTN phone lines)
- Optional VOIP service provider account (if you want to place calls via the internet)

You will need to have a good basic understanding of Windows Networking. It is recommended to read up on IP PBX's & SIP. For more information visit the IP PBX FAQ at http://www.3cx.com/PBX/IP-PBX-faq.html

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 file 3CXPHONE SYSTEM3.EXE. Click 'Next' to start installation.
2. You will be asked to review and approve the license agreement, as well as to choose an installation location. 3CX Phone System will need a minimum of approximately 50 MB of free hard disk space. You will need to reserve additional space to store voice mail files & prompts.
3. Set-up will ask you how many digits you wish your extension numbers to be. It will ask you for a preferred username and password, which you will need to logon to the management console and manage the phone system. Finally it will ask you for your mail server name and reply to address. These settings are used to send email notifications to users of new voice mail.
4. Click 'Install" to start the installation of 3CX Phone System. Setup will now copy all files and install the necessary Windows services. Click 'Finish' when ready.

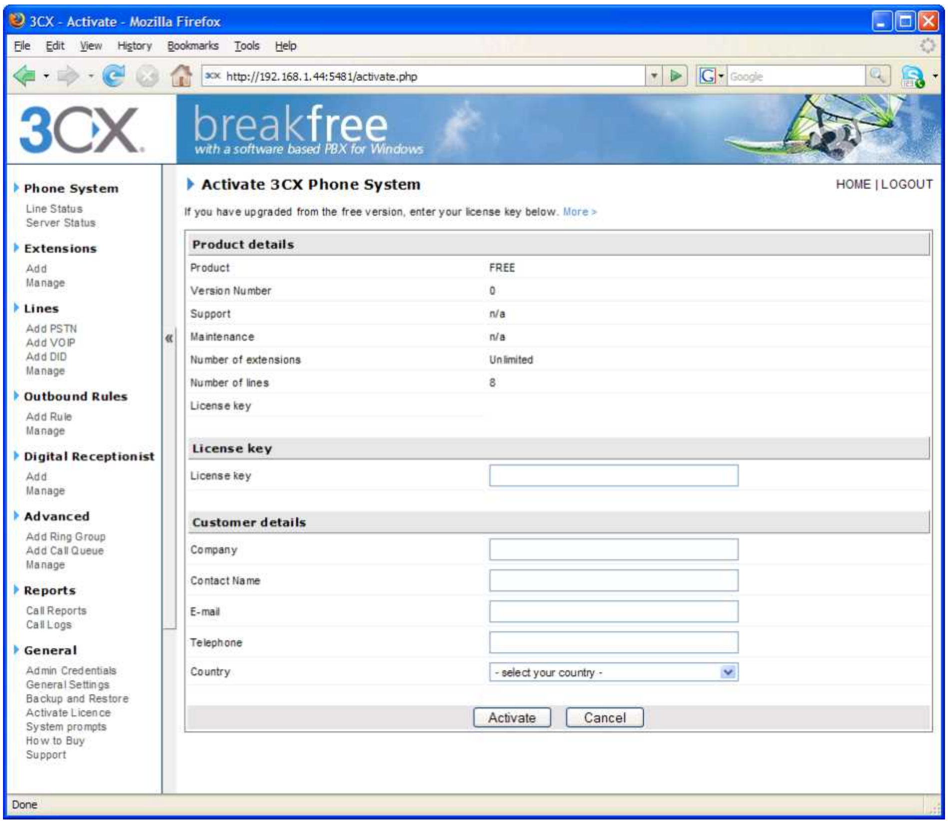
After Set-up has completed, you can connect to the 3CX Phone System Management Console by clicking on the management console short cut in the 3CX Phone System program group.

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 port 5481. (For example: http://phone-system:5481)

Note: If running Windows XP Service Pack 2 with Windows Firewall enabled, after installation, Windows XP will request confirmation to continue blocking “Apache HTTP Server” and “3CX PBX server”. Click the “Unblock” button on this dialog!

Activating 3CX Phone System

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



Screenshot 2 - Activating your license

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

Firewall configuration

There are 2 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 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)
- Port 3478 (UDP) for communication with the STUN server (send & receive)
- Port 9000-9003 (or higher) (UDP) (send & receive) for RTP communications, which contain the actual call. Each call requires 2 RTP ports, one to control to call and one for the call data. Therefore, you must open twice as many ports as you wish to support simultaneous calls via the VOIP provider. For example, if you want to allow 4 people to make calls via the VOIP provider simultaneously, you must open port 9000 to 9007.

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

Firewall installed on the 3CX Phone System machine

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

- Port 5481 (TCP) for the 3CX Phone System management console
- Port 5060 (UDP) for the SIP server
- Port 3478 (UDP) for communication with the STUN server
- Port 7000 to 7500 (UDP) for internal calls (or higher if you have more then 250 extensions).
- Port 9000-9003 (UDP) for external calls via a VOIP provider. Each call requires 2 RTP ports, one to control the call and one for the call data. Therefore, you must open twice as many ports as you wish to support simultaneous calls via the VOIP provider. For example, if you want to allow 4 people to make calls via the VOIP provider simultaneously, you must open port 9000 to 9007.

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

Installing the 3CX Call Assistant

The Call Assistant is a small windows application that allows users to manage their extension and calls from their desktop. Users can see the extension numbers of other users and whether they are available to take calls. The Call Assistant application is phone-independent, enabling you to use your favourite SIP hardware or software phone.

The Call Assistant ships with all editions of 3CX Phone System, including the Free edition. However, in the Small Business, Pro and Enterprise editions, the Call Assistant allows you to transfer calls to another extension or voice mail, place calls on hold and launch calls with a few mouse-clicks. You do not need to learn complicated phone functions.

The Call Assistant is a very small application (400k) and is very easy to deploy. To install it:

1. From the machine on which you wish to install the Call Assistant, go to: http://phone-system:5481/callassistant.php, where phone-system is the machine name of your phone system server.
2. Click on the Link 'Install 3CX Call Assistant'.
3. Click 'Run' to execute the setup process.
4. You will be prompted to select a location. The files will be copied. Click Finish to complete installation.

The 3CX Call Assistant User manual

For 3CX Phone System extension users, there is a special manual which explains the use of 3CX Call Assistant, as well as how to retrieve voice mail by phone. The latest 3CX Call Assistant manual can be downloaded here:

http://www.3cx.com/manual/3CXCallAssistantmanual31.pdf

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.

To get started, 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 on which 3CX Phone System is installed, followed by port 5481. (For example: http://phone-system:5481).

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: You can check for new translations and help text updates by selecting the “Check for other languages” option before login.

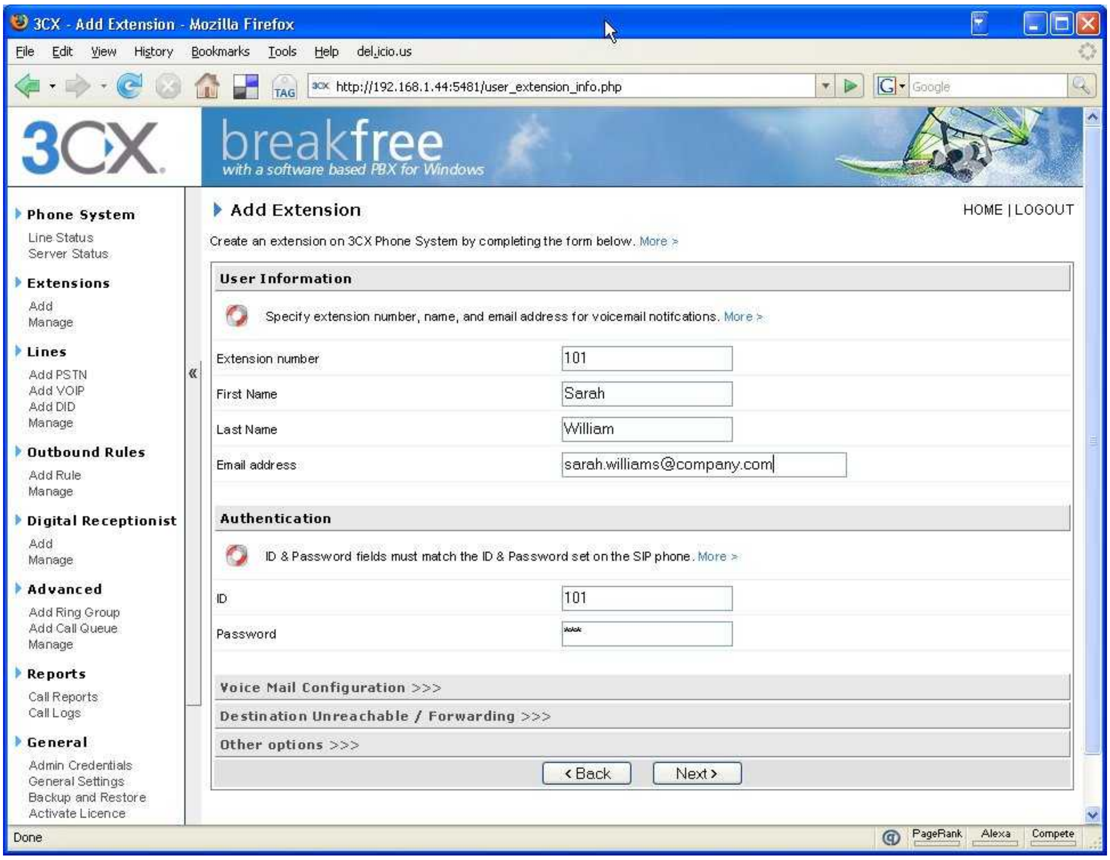
The 3CX Phone System Management console will appear. The screen is divided in 2 main sections:

- A left hand menu which includes the most important configuration sections, including Extensions, Lines, Outbound rules, Advanced and General.
- A right hand section which lists the configuration/management options for the selected configuration option.

The first time you start-up the management console it will show you a screen with the steps you still need to perform to get 3CX Phone System up and running. The second time you start-up the Management console, after you have created the extensions, it will display the line status of all extensions and external lines.

Adding Extensions

To add an extension, click ‘Add’ in the Extensions section. This will bring up the ‘Add Extension Page’.



Screenshot 3 - Adding an extension

Now enter the following information:

1. User information.

- Extension number – Specify extension number
- First name – Enter the user’s first name.
- Last name – Enter the user’s last name.
- Email address (Optional) – This will be used for voice mail notifications and as the default SIP ID. You can leave the field empty if you wish.

2. Authentication.

Specify an authentication ID and password.

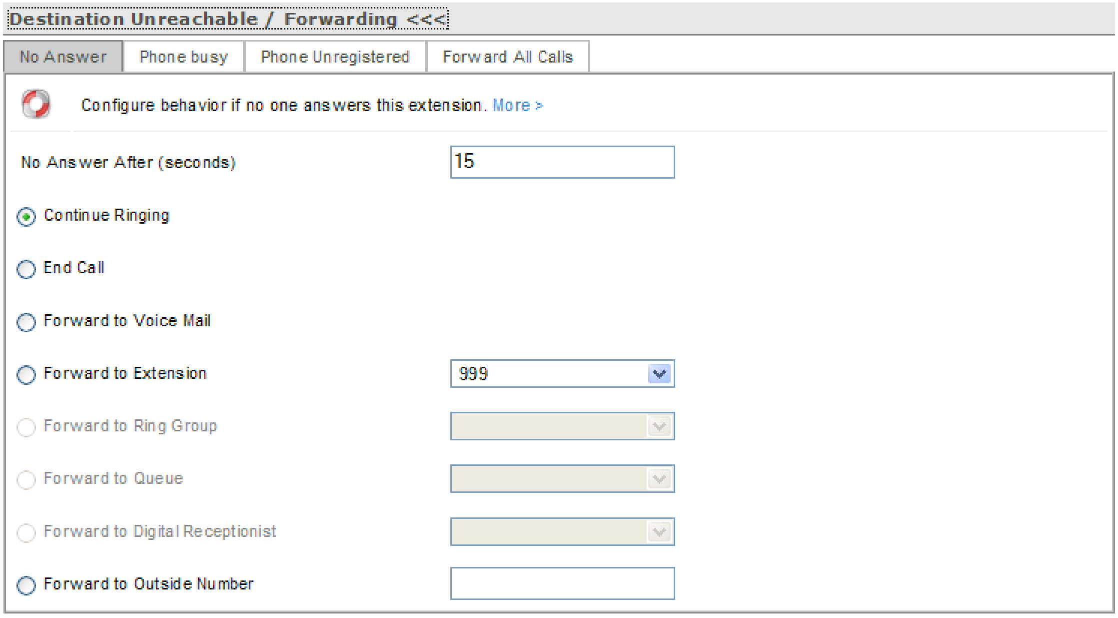
- ID – The SIP ‘Username’. e.g. 200.
- Password –The SIP Password (password can be hidden from the user).

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

3. Voice mail Configuration.

- Enable voice mail – allows you to enable voice mail for the extension/user
- 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 Call Assistant.
- 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.

Configure when calls should be sent to voice mail in the Destination Unreachable / Forwarding section of the extension. Calls can be sent to voice mail when an extension is busy, a call remains unanswered or when an extension is unregistered.



Screenshot 4 - Destination unreachable configuration

4. Destination Unreachable / Call forwarding

You can configure per extension what the phone system should do if the extension does not answer the call, is busy or is unregistered. In the case of No Answer you have to specify the time in seconds that you want the system to wait. In the case of busy, you have to specify whether you want the phone or the PBX to do the busy signaling. By default, most SIP phones do not give a busy signal back, but accept a second incoming call. If you want to redirect a call when an extension is busy, you will need to specify “Use PBX status” in the busy detection list box.

You have the following re-direction options;

- Continue ringing – the system will do nothing.
- End Call
- Forward to voice mail
- Forward to extension –Specify the extension to forward to in the list box.
- Forward to ring group – Specify the virtual extension number of the ring group in the list box
- Forward to Queue – Specify the virtual extension number of the Queue in the list box
- Forward to Digital Receptionist – Specify the virtual extension number of the digital receptionist in the list box.
- Forward to Outside number – In this case the system will forward the call to an outside number. Be sure to include the complete dial string including the prefix.

The Forward all calls option is used when a user is temporarily out of the office, for example on holiday.

5. Configure other options:

- Extension is external: If this extension is located outside the firewall, or on another subnet, specify that the extension is external. In this case the extension will be treated in much the same way as a call via a VOIP provider.
- Its possible to change the user status and queue status – these settings correspond to the settings in 3CX Call Assistant.
- SIP User ID – If you wish to allow people to reach you from any SIP phone in the world using a friendly name such as sarahjones@mycompany.com, enter your SIP ID here. SIP User ID requires additional DNS configuration.

Once you have filled out all necessary fields, click on the 'OK' button to create the new extension.

Configuring the SIP phones

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

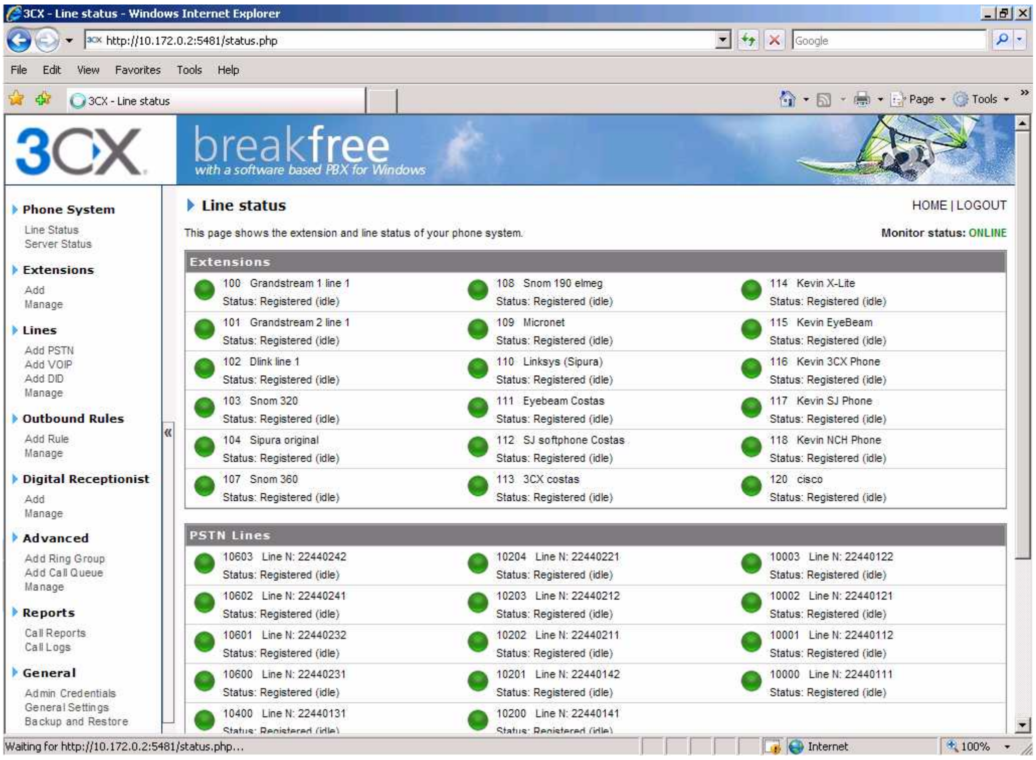
- Proxy server IP or FQDN: **Host name of 3CX Phone System**
- User ID: **Extension number created**
- Authentication ID: **As specified in Authentication ID field**
- Password: **As specified in Authentication password field**

This information needs to be entered into the SIP phone. For a detailed description how to do this for popular phones including CISCO, SNOM, SJ Phone, Aastra, Grandstream and more, see our configuration guides at: http://www.3cx.com/sip-phones/index.html

Testing your setup

As soon as you have created at least 2 extensions, and configured SIP software and/or hardware 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 5 - The Status monitor

Load up the 3CX Phone System Management console, and click on Phone System > ‘Line Status’ (This is the default page). Check that all extensions are listed and are ‘On Hook’.

The status monitor shows the status of your extensions and any external lines. It is the default screen when you load up the management console after having configured the extensions. The following states are displayed:

- **Not registered** – The extension has been created, however the phone has not registered itself with the system. This could be because the device is off, or because the SIP credentials are incorrect. 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.
- **Registered (idle)** - The extension is registered and ready for phone calls.
- **Calling** – The extension is dialling a number
- **Ringing** – The extension is ringing
- **Connected** – The extension is currently on a call
- **Holding** – The extension has placed a call on hold
- **On Hold** – The extension is party to a call that has been placed on hold

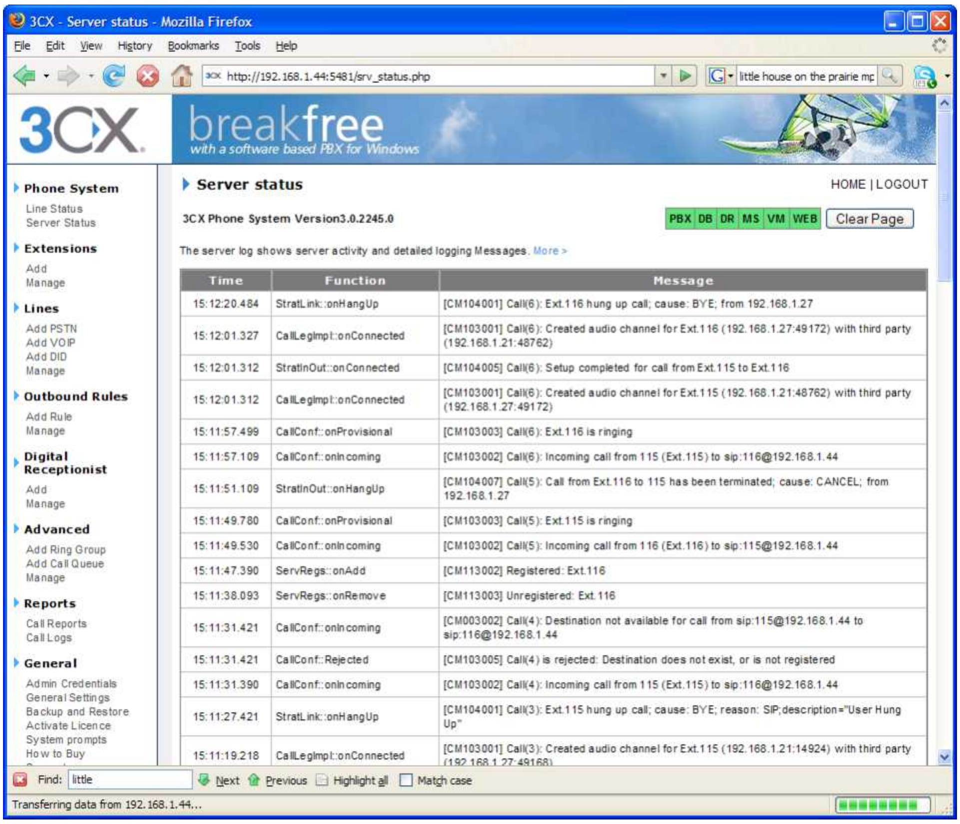
2. Place a call to another extension

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

If dialling another extension does not work, check the following:

- That you have created the extensions and configured the phones with the same authentication details and that they have registered. If they have not registered, you have most likely entered the Extension Number, Authentication ID and Authentication Password in the wrong fields.



Screenshot 6 - 3CX Phone System server status

- Click on **Phone System > Server status** to bring up the server status screen. This screen shows the activity log of the server. Often these messages can give you further information on what the problem is. A detailed description how to understand the server status messages can be found in the support section of the 3CX website.
- Check your Firewall configuration! Switch off the firewall temporarily to check if the problem is caused by the firewall.

Importing extensions

If you need to create a large number of extensions it is handier to bulk import the extensions. To do so follow these steps;

1. Create a spreadsheet with columns for each field. The latest list of fields can be found in the “Import extensions screen”

2. Now list all the extension numbers, user names, authentication ID and password and other optional fields on separate lines. After you have entered all the desired extensions, save the file as a “CSV (Comma Delimited)” file.

4. In the 3CX Phone System Management Console, go to the “Manage” page under the “Extensions” section. Click on the “Import Extensions” icon to load up the ‘Import extensions’ page. Click ‘Browse’ and select the file previously created. Click on the ‘Import’ button to import and create the extensions.

4. 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.
- 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. This chapter explains VoIP Gateways and how to configure them.

What is a VoIP Gateway?

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

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

VOIP Gateway 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 to register its lines with 3CX Phone System. Each line gets a SIP user ID & Password, and 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.

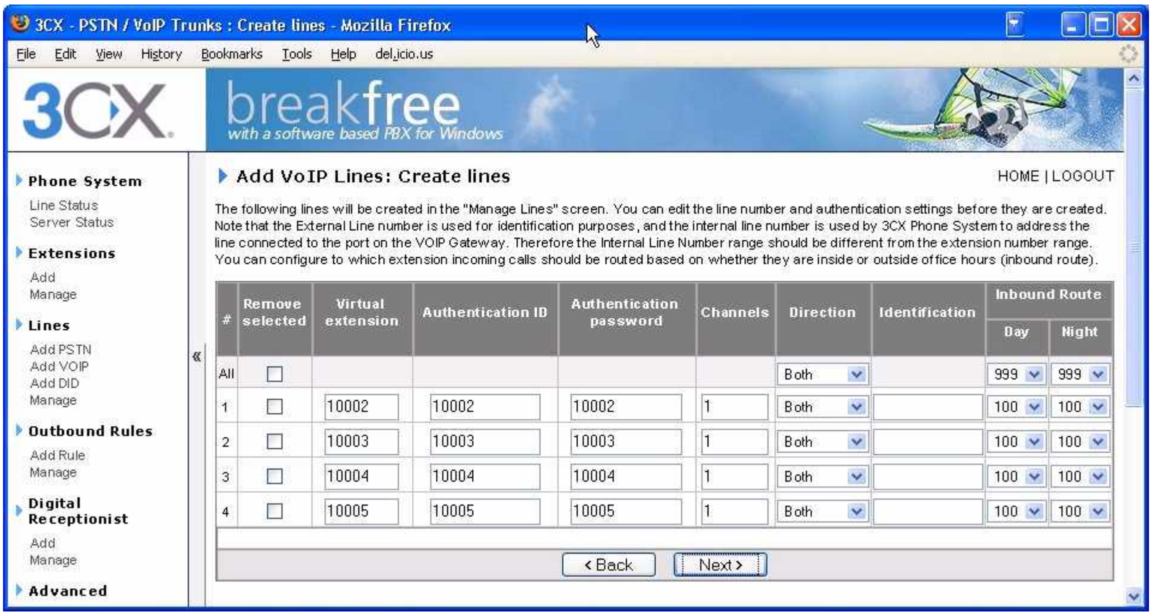
Step1: Create the PSTN lines in the 3CX Phone System

After you have decided for a particular VoIP Gateway model and obtained the device, the first step is to create the PSTN lines in the 3CX management console. Each PSTN Line will become a virtual extension and be assigned a virtual extension number, authentication ID and authentication password. To add PSTN lines to the 3CX phone System;

1. In the 3CX Phone Management console click on the ‘Add PSTN’ link in the left hand menu.
2. In the name field, enter a friendly name for the PSTN gateway. Now choose the gateway 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. 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.

Screenshot 7 - Add VOIP Gateway

4. Specify the number of ports that the gateway supports and whether you are using Analog lines, BRI, PRI or T1 lines. This will set up one account for each port and enable the corresponding number of 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 lines. If you wish to have each line individually addressed, simply select 23 * Analog lines.
5. 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 8 - Adding PSTN Lines

6. Click Next. The individual lines will be 'created' and displayed in a columnar format.

Virtual extension – In effect the VOIP Gateway "converts" each line 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 phone lines connected to the VOIP Gateway 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.

Direction: This field specifies whether the port will be used for inbound calls, outbound calls or both. By default it supports both 'directions'.

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

Inbound Route: If the line/port will receive inbound calls, you can specify to which extension, ring group or digital receptionist a call must be routed. You can specify a different option depending on whether the call is received inside or outside office hours.

7. When done click finish to create the lines. You can change line options of lines that have been created by going to the Lines > Manage page.

8. On the next page, you will be asked for a prefix so as to create an outbound rule for this device. Tick the "Create outbound rule" check box and enter the dialling prefix in the "With prefix" text box. To make calls via this provider, precede the number to be dialled with this prefix.

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 line/port that you have on your gateway.

For a detailed description how to do this for popular gateways including Patton, Grandstream and Audiocodes and more, see our configuration guides at: 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 configuration' 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.

5. 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 25k-64k 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.

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. You can find the latest list of supported VOIP providers at:

http://www.3cx.com/support/voip-providers.html

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.

Note that Skype has a proprietary protocol and is therefore currently not supported.

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 click on the 'Add VoIP' link in the Lines section in the left hand menu.
2. In the name field, 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'. In that case we will not able to guarantee that it will work with this VOIP provider. Now click Next.

Screenshot 9 - Add VOIP Provider account

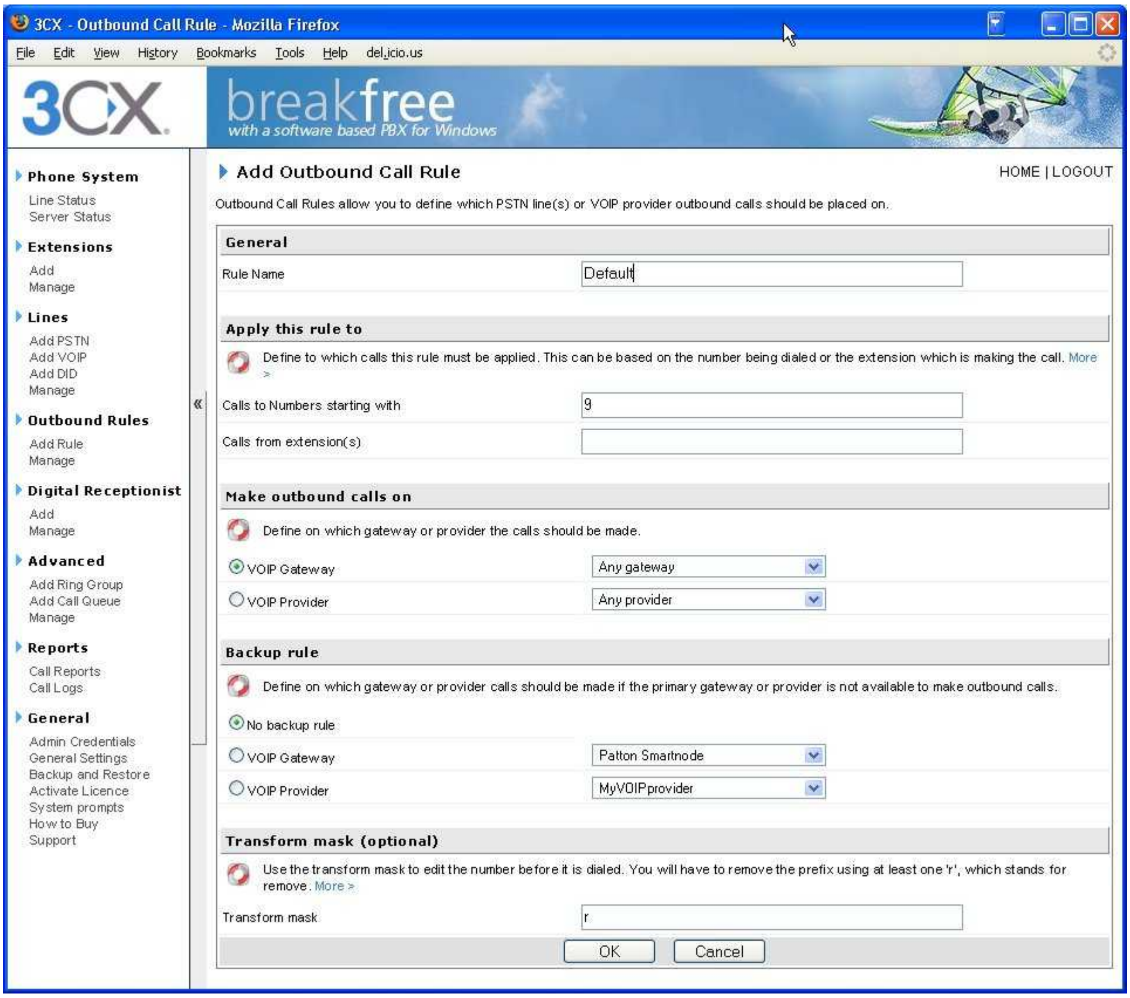
3. The registrar/Proxy Hostname will be pre-filled. Compare these with the details that you have received from your VOIP provider and check that these are indeed correct. Depending on the VOIP provider that you are using, some fields will be disabled. This means you do not need to change them. You do not need to change any of the options included in the other sections. 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. You can configure how inbound calls should be routed in the 'Route inbound calls section'. You do not need to change any of the other options included in the other sections. Click Next to continue.
5. On the next page, you will be asked for a prefix so as to create an outbound rule for this VOIP provider. Tick the "Create outbound rule" check box and enter the dialling prefix in the "With prefix" text box. To make calls via this provider, precede the number to be dialled with this prefix.

6. Creating Outbound Call Rules

Introduction

An outbound rule defines on which line/gateway/provider an outbound call should be placed based on who is making a call and who he/she is calling.

Creating an outbound call rule



Screenshot 10 - Creating a new outbound rule

To create an outbound rule:

1. Click on 'Add Rule' in the Outbound rules section of the 3CX Management console. Give a name to the rule in the general field.

2. Now specify to what calls to apply the outbound route. In the 'Apply this rule to these calls' section, specify either or both of these options:

- Calls to Numbers starting with – apply this rule to all calls starting with the number you specify. For example, specify 0044 (or 01144 in the US) to specify that all calls to the UK should be routed via a particular line or service provider
- 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

3. Now specify how the outbound calls should be made. In the 'Make outbound calls on' section, select:

- VOIP Gateway – select the VOIP Gateway on which the calls should be made.
- VOIP Provider – select the VOIP provider on which the calls should be placed.

4. You can define a backup rule, just in case the primary provider or gateway is not available or is busy.

5. You can transform a number that has been 'caught' by an outbound rule before it is routed to a selected gateway or provider with the use of a 'transform mask'. The following are valid 'transform mask' components.

"r" "r" can be used to remove an integer from the dialled number. Several r token's can be used to remove several integers. e.g. [r] removes the leading 0 in [00131584210212] to produce [0131584210212], [rrr] would remove the first 3 0s in [00131584210212] to produce [31584210212],

INTEGER The use of an integer in a transform mask ADDS the integer to the dialled number. e.g. [001] would transform a caught number [31584210212] to produce [00131584210212]

Manage outbound call rules

To manage outbound rules, click on Outbound Rules > Manage in the 3CX Management console. This brings up the list of the rules and shows you at a glance how outbound calls are routed. You can configure the priority of a rule by moving the rules up or down in the list.

Examples of Outbound Rules

Routing calls by destination prefix example

This example illustrates how calls are routed to Gateways or providers according to dialled number prefixes.

☐	Name	Apply for Extensions	Number prefix	Transform mask	Make calls on
☐	Foreign_US/Canada/UKCalls		001,0044		VoIP Provider: "VP_voipbuster"
☐	Foreign_GermanyCalls		0049	rr	VoIP Provider: "VP_nikotel"
☐	Local_LandLineCalls		020-030	r	PSTN Gateway: "PSTNGW_001"
☑	Local_CellularCalls		079,099	r	PSTN Gateway: "PSTNGW_002"

Screenshot 11 - Outbound call rule example

Here the first rule will catch numbers dialled with the 001 (Canada & US) or 0044 (UK) prefix, and place them (Canada, US or UK calls) on the 'VP_voipbuster' VoIP Provider line.

The second rule will place calls bound to 0049 (Germany) (with additional 0) on the 'VP_nikotel' VoIP Provider line. The number will be stripped of its leading 0s. This is useful since Nikotel works without the 00s in the dialled number's prefix.

The third rule in the example catches all calls with that start with a number that start with 020 to 030 (i.e.; 020,021,022,023,024,025,026,027,028,029,030), the calls are then placed to the "PSTNGW_001" Gateway. The number will be stripped of its leading 0.

The forth rule catches only numbers with 079 or 099 and places them on the second gateway.

Examples:

Dialled: 00131588008840	Transformed: (same)	ON: 'VP_voipbuster'
Dialled: 0049588008866	Transformed: 49588008866	ON: 'VP_nikotel'
Dialled: 020880088	Transformed: 20880088	ON: 'PSTNGW_001'
Dialled: 022880088	Transformed: 22880088	ON: 'PSTNGW_001'
Dialled: 079880088	Transformed: 79880088	ON: 'PSTNGW_002'
Dialled: 099880088	Transformed: 99880088	ON: 'PSTNGW_002'

Routing calls by dedicated line prefix example.

	Name	Apply for Extensions	Number prefix	Transform mask	Make calls on
☐	#01 - InternationalCalls		01	rr	VoIP Provider: "Nikotel"
☐	#02 - CellularCalls		02	rr	PSTN Gateway: "PSTNGW_001"
☐	#03 - LandLineCalls		03	rr	PSTN Gateway: "PSTNGW_002"

	Name	Apply for Extensions	Number prefix	Transform mask	Make calls on
☐	#01 - InternationalCalls		1	r	VoIP Provider: "VP_nikotel"
☐	#02 - CellularCalls		2	r	PSTN Gateway: "GW_GXP4104"
☐	#03 - LandLineCalls		3	r	PSTN Gateway: "GW_MN_SP-5050"

Screenshot 12 - Outbound call rule example

Instead of catching calls by prefix, you can dedicate a prefix to a certain provider and use that prefix in order to route calls to a desired provider. The prefix would need to be stripped before being routed to the provider. In the above examples this is done with the [r] 'Transform mask'.

Examples:

Dialled: 100131588008840	Transformed: 00131588008840	ON: 'VP_nikotel'
Dialled: 200131588008840	Transformed: 00131588008840	ON: 'GW_GXP4104'

7. 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 or mp3 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.

You can have 2 options for creating prompts:

1. Record the prompt directly in Windows Sound Recorder

2. Use a text to speech program or website to create the prompt.

Recording the prompt in Windows Sound Recorder

To record the prompt in Windows Sound Recorder:

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

Screenshot 13 - Windows Sound Recorder

2. Now click the record button to record your system prompt.

3. Enter a name for the prompt and save the prompt. When creating a digital receptionist menu you will be asked for the path to the file.

Creating a prompt using Text to Speech

You can also create a prompt using a text to speech utility. There are several good text to speech utilities. You can either download a program such as Balabolka (http://www.cross-plus-a.com/balabolka.htm) or you can use an online text to speech converter, for example

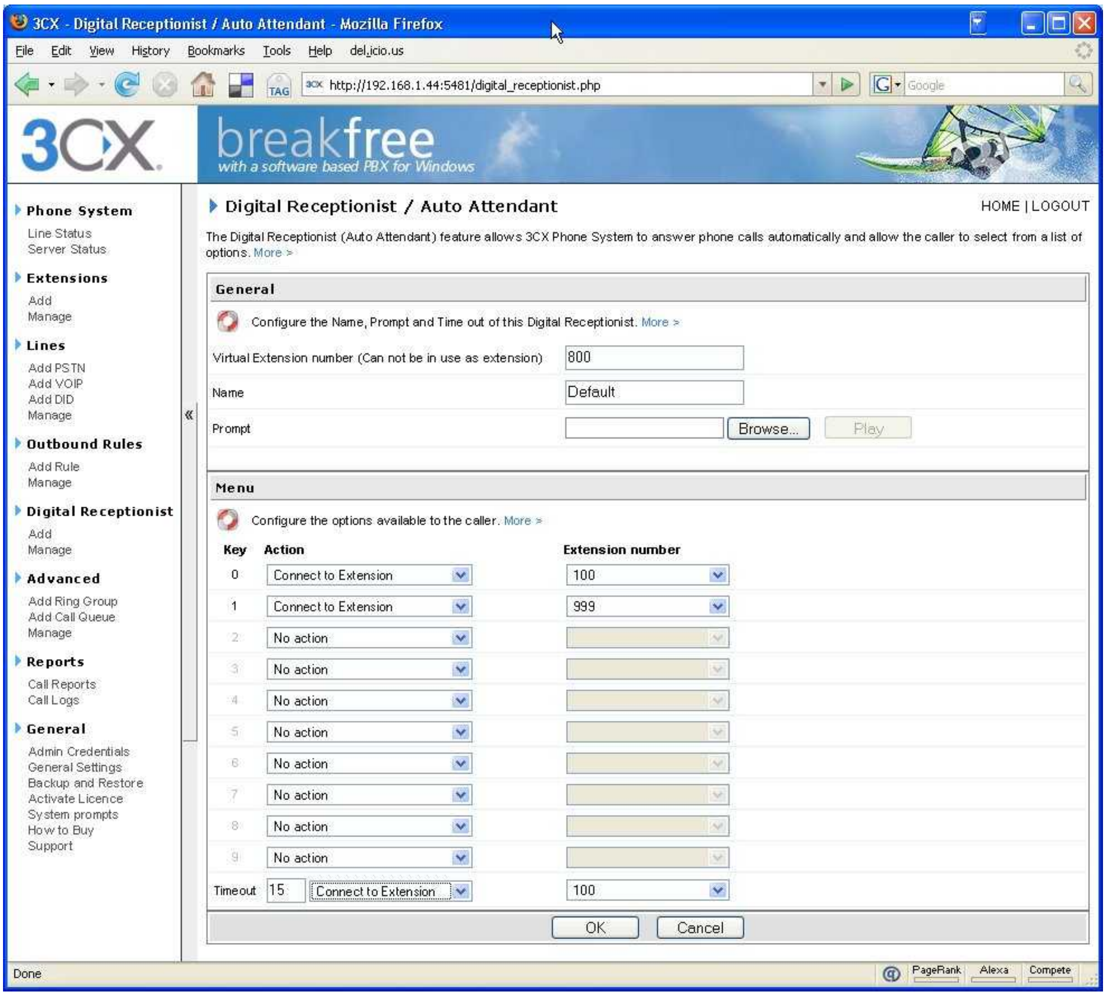
http://www.research.att.com/~ttsweb/tts/demo.php

You can purchase text to speech libraries for particular voices or languages at http://www.nextup.com/. You can also have voice prompts professionally recorded using a 'Voice Talent' studio.

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 Management console click on the 'Add Receptionist' link in the left hand menu. The 'Digital Receptionist' screen appears.



Screenshot 14 - Configuring a digital receptionist

2. Specify a name and virtual extension number for the digital receptionist.

3. Now click on the browse button and specify the file that you previously recorded. The file will be copied into the 3CXPhone System\Data\Http\prompts\ directory.

4. Now you can 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 can automatically perform an action. This is handy for callers who did not understand the menu or who do not have a DTMF capable phone. When you are 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. This allows callers who know their parties extension to connect to their party without having to listen or to bother the receptionist and without requiring you to setup an option for each extension in your company.

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”

8. Adding Direct numbers using DID/DDI lines

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. VOIP providers also provide DID lines. Enquire with your Phone Company or VOIP provider for more information.

DID numbers and 3CX Phone System – How it works

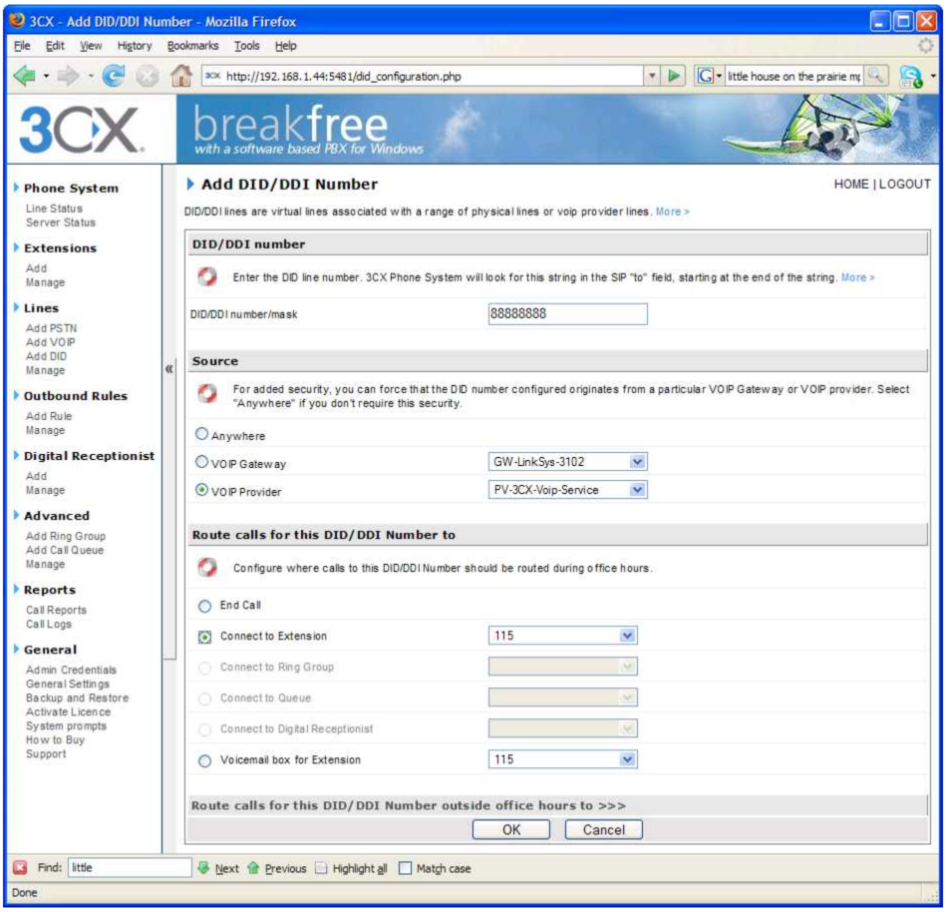
DID numbers “play well” with 3CX Phone System, or indeed with any SIP based IP PBX. Acting according to the SIP standard, 3CX Phone System expects the DID number, i.e. the intended destination, to be in the To: field of the SIP invite request. Most VOIP providers and VOIP Gateways will do this by default.

If your VOIP Provider and VOIP Gateway do this by default, all you need to do in the 3CX Management Console is setup a DID Line (which is akin to an inbound rule) to configure calls made to that particular DID number to go to a particular extension, digital receptionist or other destination.

Adding a DID line

To add a DID line to 3CX phone System;

1. In the 3CX Phone Management console click on the ‘Add DID’ link in the left hand menu. The ‘DID/DDI configuration’ screen appears.



Screenshot 15 - Adding a DID Line

1. 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.

   Another way of finding out the exact number that is inserted in the “To” field is by viewing the server status log and seeing the “To” field assigned by the VoIP Provider / Gateway and then adjusting the DID line number accordingly.

3. In section “Source”, select the Gateway or Provider which provides the DID number (i.e. from which the call will originate). This field is optional but adds a level of security, since otherwise 3CX Phone System will accept calls to that DID number from any source. If you are not using a VOIP provider but only a VOIP Gateway, you could leave this field unchecked.

4. In the next section you can specify where calls to this DID line should be routed to:

Route inbound calls in office hours:

This section allows you to specify how an inbound call should be routed during office hours. You can choose to route inbound calls to a:

- Digital receptionist – Select the digital receptionist from the drop down list of configured digital receptionists (note that digital receptionists are not available in the beta version).
- Extension – Select the extension from the drop down list of available extensions.
- Ring group – Select the ring group from the list of available ring groups.
- Queue – Select the queue from the drop down list of available queues (Enterprise edition).

Route inbound calls out of office hours:

You can specify that an incoming call is routed differently if it is received outside office hours. You can select from the same routing options. Click on the 'Specify Office Hours' button to define your office opening hours.

Troubleshooting DID lines

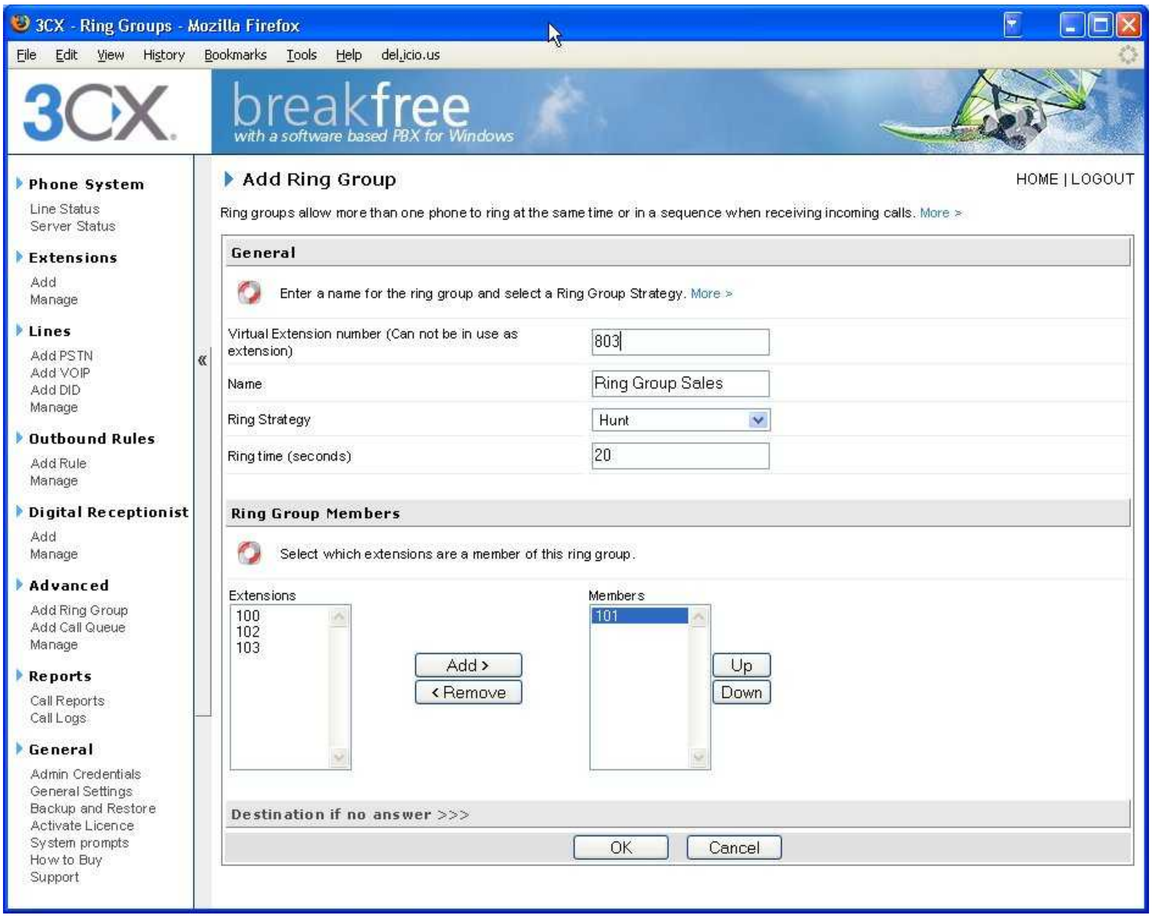
If you have created DID lines, but calls are not being forwarded, its best to do the following:

1. Go to the Server Status screen 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

9. Ring groups & Call Queues

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 16 - Adding a ring group

To add a ring group:

1. Click on the Advanced > Add Ring Group link 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 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

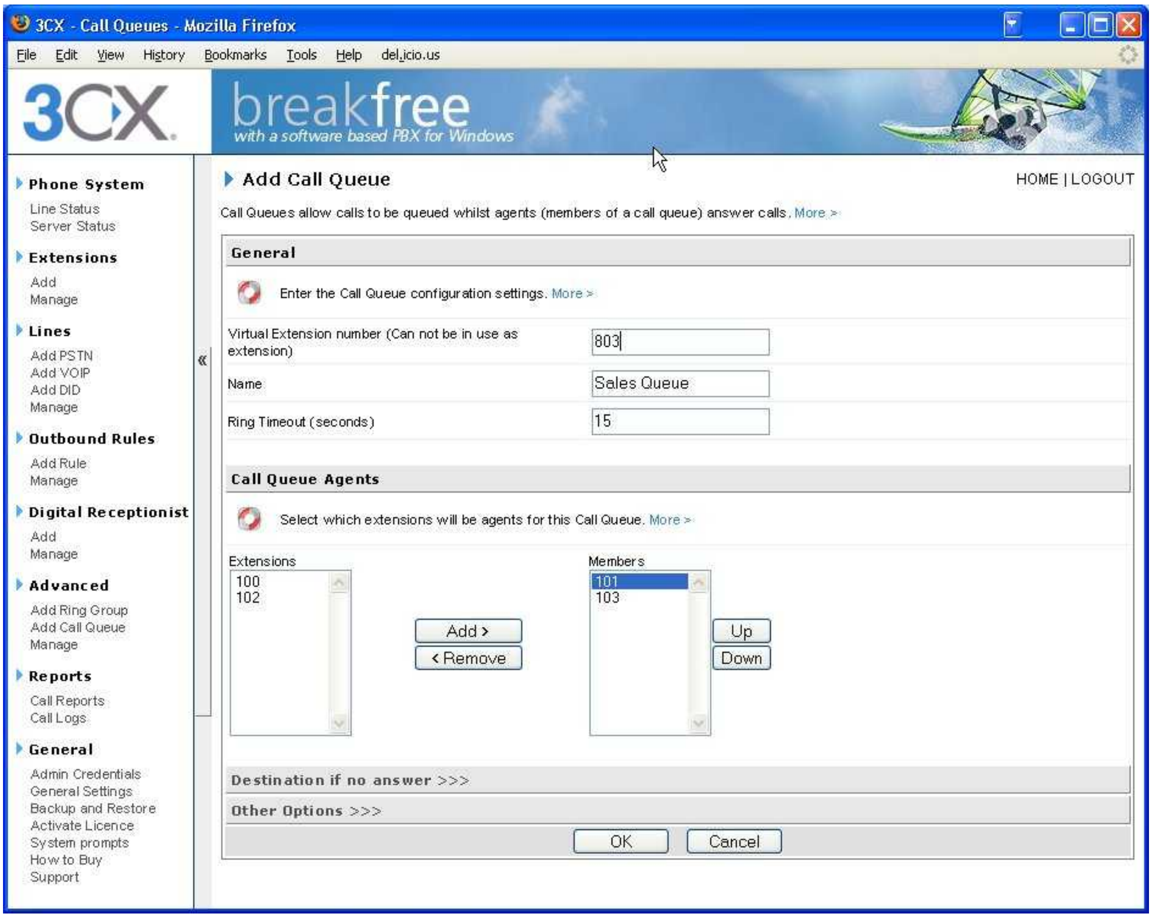
- Ring strategy – Select the appropriate ring strategy for this ring group:
    - o Hunt – this will start ringing on the first extension, then the second etc.
    - o Ring all – all phones will ring at the same time
- Ring time – Specify how long the phones should ring for.

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.

Call Queues (Enterprise edition only)

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



Screenshot 17 - Adding a Call Queue

To add a Call Queue:

1. Click on the Advanced > Add Call Queue link to bring up the ‘Add Call Queue’ page.

2. Now enter the call queue 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 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 Call Assistant.

4. In the section ‘Destination if no answer’, you can define what should happen if the call does not get answered by the Call queue.

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.

10. Generating Call Reports

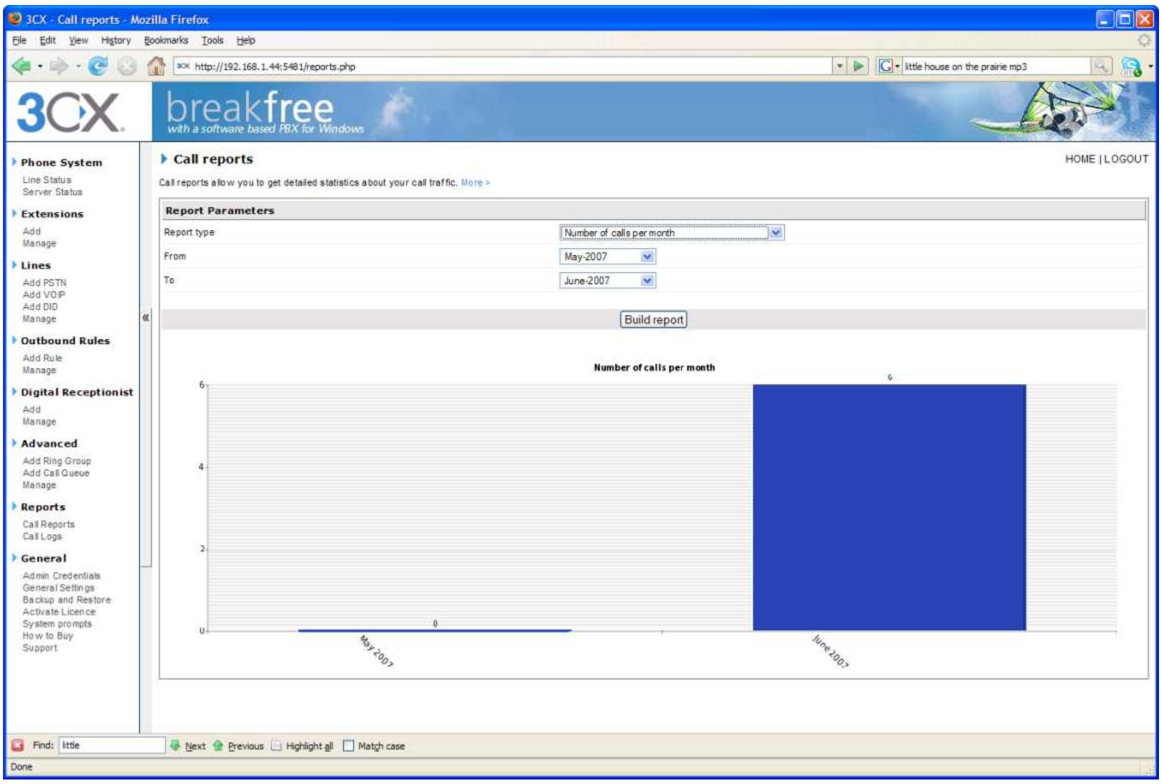
Introduction to reports

The reports feature allows you to analyze your call traffic. You can generate reports on your call traffic using the call reports feature or analyze the logs directly using the call logs feature. Using this feature it is also possible to export the call data in CSV format so you can import them into a spreadsheet or database program for further analysis.

Creating a report

To create a report:

1. In the 3CX Phone Management console click on the 'Call Reports' link in the left hand menu. The 'Call Reports' screen appears.



Screenshot 18 - Number of Calls Report

2. Now select the type of report type you want. You can select from the following reports:

- Number of calls per month
- Number of call minutes per month
- Number of outbound calls
- Number of outbound call minutes per month

- o Number of inbound calls per month
- o Number of inbound call minutes per month
- o Number of calls per extension
- o Number of call minutes per extension
- o Number of calls per ring group
- o Number of call minutes per ring group
- o Number of calls per gateway/provider
- o Number of call minutes per gateway/provider

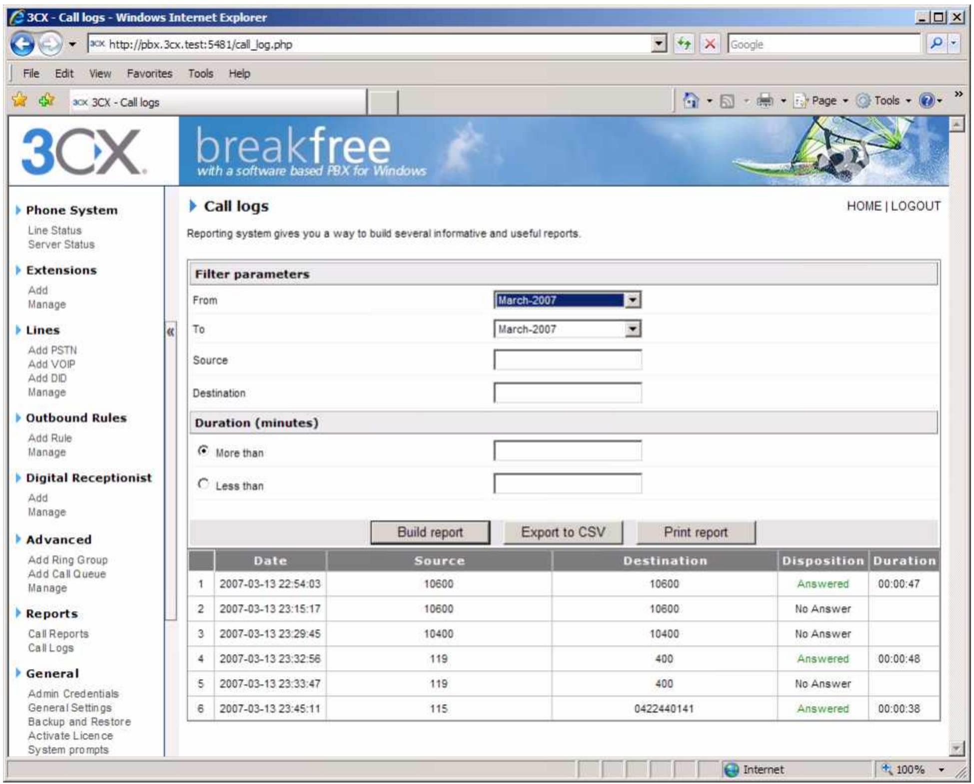
3. Now select for which months you want the report.

4. Click 'Build report' to generate the report.

Analyzing the call logs

If you need to look at the individual calls, for example to find times and destinations of particular calls or to see the call traffic for a particular extension, use the Call log feature. To create a report using the call log feature:

1. In the 3CX Phone Management console click on the 'Call Logs' link in the left hand menu. The 'Call Logs' screen appears.



Screenshot 19 - Analyzing the call logs

2. Specify the time frame that you wish to analyze by indicating a starting month in the "From" field and an ending month in the "To" field.

3. You can specify a source and/or destination filter. For example, entering an extension number in the source field will only show the calls made from that extension.

4. Optionally you can filter calls that were longer than or less than a particular time. Specify the time in minutes.

5. Click on the ‘Build report’ button to build a report in table format. Click on ‘Export to CSV’ to export the report to a CSV file (Comma Separated Values). This file can then be imported in Microsoft Excel or other products for further analysis. Click on ‘Print Report’ to print the report directly.

11. General configuration

Introduction

This chapter describes general configuration options of 3CX Phone System.

Admin credentials

To access the 3CX Management console, a user requires credentials. You specified these credentials during set-up. You can change these credentials as follows:

1. Click on **General > Administrator credentials,** to bring up the 'Admin credentials' page.

2. Now enter the new Administrator ID and password. Re-enter the password in the section 'Confirm Administrator Password'.

3. Click OK to change the user ID and/or password. At the next logon to the Management console you will have to specify the new ID and password.

General Settings

The general settings page allows you to configure a number of important options, including the ports range to be used by 3CX Phone System.

SIP Port

You can customise the port on which the 3CX Phone System will be listening for SIP messages. The default port number is 5060.

Ports to use for internal calls

In this section you can enter the ports that will be used by 3CX Phone System for internal calls. These ports will be used for calls between extensions and to external numbers routed via a VOIP Gateway. (Since your VOIP Gateway will be located on your internal network, calls via the VOIP gateway are considered internal calls).

Ports to use for external calls

This section allows you to configure which ports will be used for external calls **via a VoIP provider**. The range you specify here must be opened on your firewall. For more information see the section 'Firewall configuration' in Chapter 2.

Voicemail settings

The voice mail settings section allows you to configure an SMTP mail server and a reply to address. This is needed in order for 3CX Phone System to send out email notifications when a new voice mail arrives for a user. Optionally 3CX Phone System can attach the voice mail as a wav file. Specify the SMTP server name or IP address and a reply to address to be used as the sender name. If your SMTP server requires authentication, enter a user name and password to use.

Special Voice Menu Extension Number: This is the number to dial for the retrieval of Voicemail. The user will be prompted to dial his extension number and pin number before obtaining access to his Voicemail.

STUN Server Options

You can specify a STUN server to use for external IP address resolution. You can optionally specify a custom port. The default port number is 3478. You may also adjust the interval between one check and another. The default interval is 1200 seconds (20 minutes).

Settings for direct SIP calls

3CX Phone System includes the ability to make and receive Direct SIP Calls. 3CX Phone System can receive Direct SIP Calls (user1@mydomain.com) as well as place Direct SIP calls to Uri's from another domain. This feature requires setting up a DNS SIP SRV record for your domain to receive inbound Direct SIP Calls. For information how to do this, check this FAQ at http://www.3cx.com/support/sip-dns-configuration.html

To enable this feature, tick the 'Allow Calls to External SIP URI's' box and enter the local SIP domain you configured.

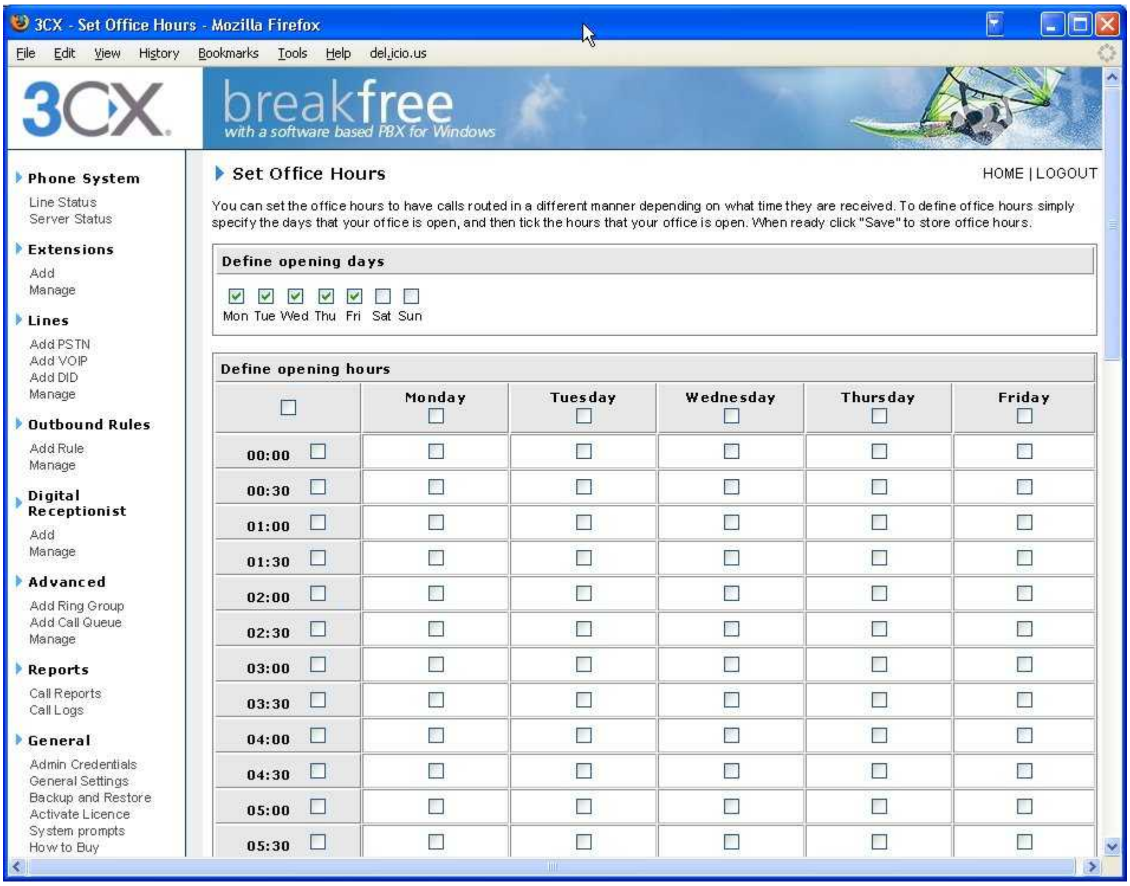
Logging Level

These settings allow you to configure the level of logging detail produced by the 3CX Phone System logging system. It is recommended to keep this setting to "Low" during normal day-to-day operations. The "Verbose" setting is a valuable tool when troubleshooting issues, but generates a large amount of data, putting greater demands on the I/O subsystem and consuming considerable storage space.

Configuring office hours

If the phone system knows your normal office hours, it can route a call differently depending on whether the call is received within or outside office hours. To configure office hours:

1. Go to the Lines > Manage page and click on the 'Office hours' button. This brings up the 'Set Office hours' page. Here you can define the days that your office is open and the daily opening times.



Screenshot 20 - Defining Office Hours

2. Now tick the days that your office is open. Those days of the week will appear in the table. Then tick the hours that your office is open. Click 'Save' when done.

Other

Music file on hold: Here you can specify a wav file with the music that you wish to play for callers that are on hold.

Exchange 2007 Unified Messaging Integration

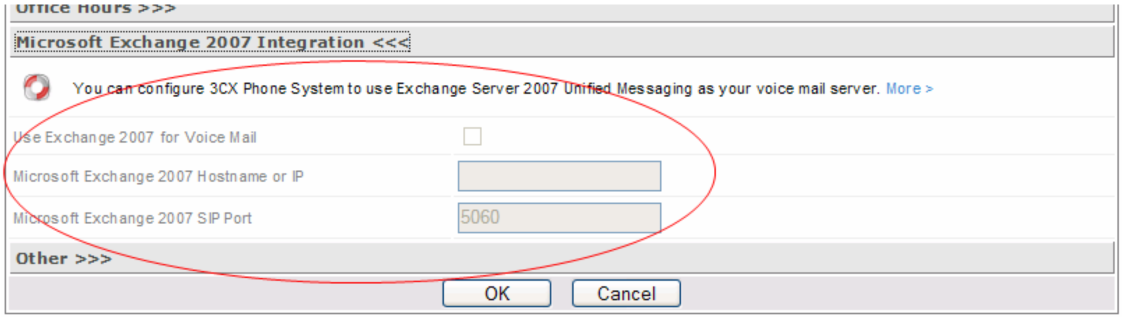
Although 3CX Phone System includes its own voice mail server and auto attendant, you can configure 3CX Phone System to use the Exchange Server 2007 Unified Messaging Server as a Voice mail server instead.

To use Exchange Server 2007 as a voice mail server, you will need to configure Exchange Server 2007 as a Unified Messaging Server. For a detailed description how to do this read our online configuration guide at:

http://www.3cx.com/support/exchange-server-2007.html

After that, you will have to enable the MS Exchange 2007 integration from the 3CX Management Console:

1. Ensure you have an Enterprise Edition of 3CX Phone System and that you have activated the license. Otherwise the fields will be greyed out.



Screenshot 21- Exchange 2007 integration

2. Go to General settings > Microsoft Exchange Server 2007 integration. Enable the feature, and enter the IP of the machine running the Exchange Server 2007 Unified Messaging Role. If you have configured Exchange Server 2007 to run on a different port, specify the port you used. Click OK.

From now onwards, the Retrieve voice mail extension, 999, will be redirected to the Exchange server Menu. All Voice mail messages will be stored in Exchange.

System prompts

System prompts are messages played to the caller by the phone system for standard situations within the voice mail and digital receptionist menus. For example, when the caller has listened to all his voice mail messages the system will play a system prompt that tells the caller that there are no more messages.

3CX Phone System ships with a default set of English system prompts. To modify the System Prompts, go to the 'General > System Prompts' page.

You can download prompts for other languages by clicking on the 'Check for more prompt sets' button – this will check if additional prompt sets are available and download them for you. You can also re-record the system prompts yourself if you prefer.

You can copy an entire prompts set and make changes to particular system prompts whilst keeping a backup of the original set. To do this, simply click 'Create Copy' and enter a new name for the prompts set.

In the prompts section all system prompts are listed, with their particular meaning and a path to the file. You can replace a particular file simply by recording the new prompt and then selecting the new prompt using the 'Browse' button.

Backup and Restore

3CX Phone System includes a convenient backup and restore function, that allows you to create a complete backup of your phone system configuration to a file.

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.

12. Troubleshooting

Introduction

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

- The manual – most issues can be solved by reading the manual.
- The 3CX FAQ – http://www.3cx.com/support/index.html
- The Phone configuration guides – http://www.3cx.com/sip-phones/index.html
- The Gateway configuration guides - http://www.3cx.com/voip-gateways/index.html
- The Support forum – http://www.3cx.com/forums/
- Our support system

Manual

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

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

Configuration guides

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

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

FAQ

3CX maintains an FAQ, which includes answers to the most common problems. If you have a problem, please consult the FAQ first. The FAQ can be found at http://www.3cx.com/support/index.html

Web Forum

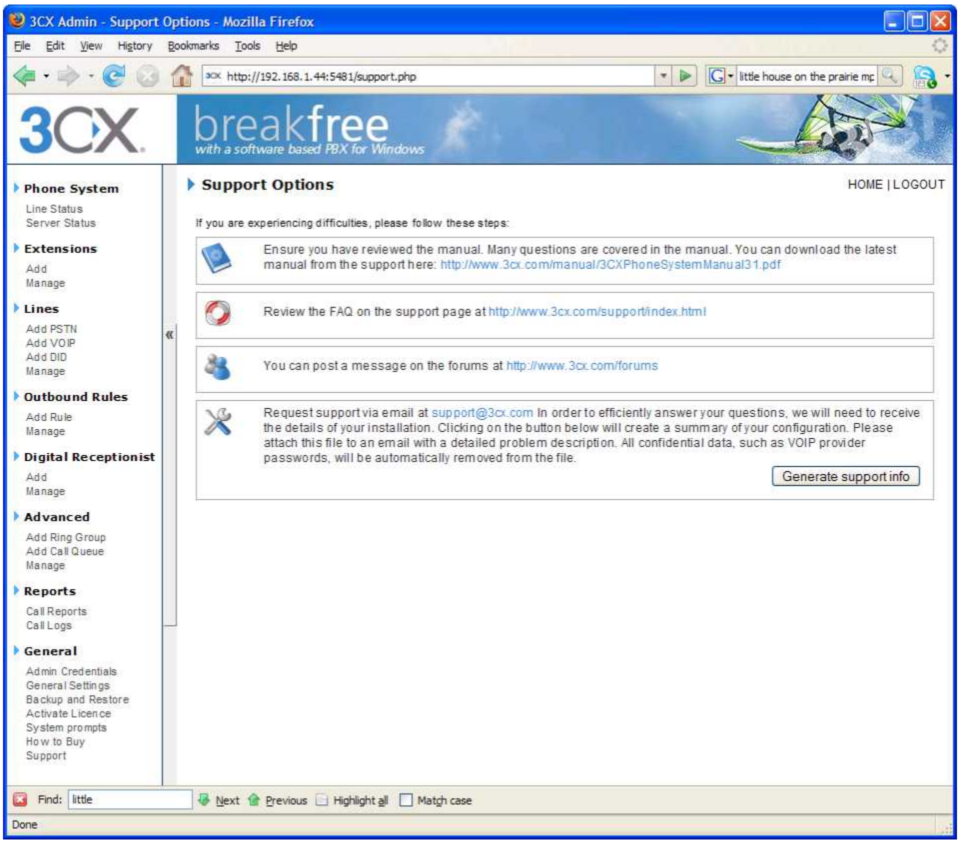
User to user support is available via our web forums. The forum can be found at:

http://www.3cx.com/forums/

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.



Screenshot 22 - Generating support information

To generate the support info file:

1. In the 3CX Phone System Management console, click on **General > Support.** This will load up the 'Support options' page.
2. Click on the button 'Generate support info' in the bottom section.
3. You will be prompted for a location to save the data. 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.

Index

Extensions 19
Importing extensions 25
media server 13
office hours 29, 43, 54
outbound route 33
PSTN Interface 30
PSTN lines 27
SIP phones 11
Skype 31
Troubleshooting tips 25
VOIP Gateway 27