PBXware System Administration Manual

Part I

1 Introduction

PBXware

PBXware is a scalable telephony solution featuring a range of traditional telephony and emerging VoIP technologies. The creation of a national/global voice networks in addition to a complete range of Analog, Digital, and VoIP channels, all fully supported. Functionality includes: Voicemail, IVR, ACD Queues, Call recording/monitoring, Conferences, Auto Provisioning/Configuration and much more.

Part II

2 Getting Started

In this chapter we will cover:

2.1 System Requirements

What is needed to get the PBXware up and running?

Server Hardware

- Standard x86-compatible server hardware
- 400MHz or faster CPU minimum (2000 MHz recommended)
- Compatible processors include: Intel®: Celeron®, Pentium® II, Pentium III, Pentium 4, Xeon™, AMD™: Athlon™, Athlon MP, Athlon XP, Duron™, Opteron™, Via C3, C7

Memory

- 128 MB minimum (256 MB or more recommended)

Display

- None required. (Exception is during the installation only for)

Disk Drives

- Standard IDE or SCSI hard drives (CD ROM, Tarball installs only)
- Free space for installation—2GB for
  CD ROM Installation Method=20GB
  Compact Flash Installation Method = 20GB (Please note: This does not mean the compact flash must of this size. This size applies to hard disk where the image from compact flash will be copied to)
  USB Disk =
- Standard CD-ROM/DVD-ROM drive (CD ROM Install only)

Local Area Networking

- Any Ethernet controller supported by the operating system
- Network configured and fully setup with DHCP service

Optional:

- If the server is to operate from a private LAN IP then the firewall must be opened to the following ports: **TCP:** 80, 81, 443, 10001, 5038, 5060-5069, **UDP:** 4569, 5060-5069, 10000-20000

Linux Operating System

One of the following Linux operating systems is required (tarball edition ONLY)

- Red Hat Enterprise Linux (RHEL) 2.1, 3.0. 4.0
- Cent OS 3.0, 4.0
- Mandrake Linux 9.0, 10
- Gentoo 2004, 2005
- Red Hat Linux 7.2, 8.0, 9.0
- SUSE Linux 8.0, 9.0
- Debian 3.0
- Fedora (Core 3)

NOTE:
1. Operating system and all other necessary applications are supplied with PBXware CD ROM and appliance delivery methods..
2. If ZAPTEL hardware is to be used, it should be inserted into the server before installation as to ensure proper loading of the modules
3. X is not required
4. Any of the above distributions need to be installed with their default "server" installation making sure that bison, openssl and openssl-dev or libssl-dev, libnewt are installed
5. Kernel should be Linux 2.6.x+ with kernel sources and headers available.

Supported Browsers
System can be administered by using one of the following web browsers:
- Internet explorer 6.0+
- Firefox 1.0+
- Opera 9.*+

Support Requirements
In order to provide systems support we need from time to time access to the system server by SSH, HTTP/HTTPS protocols the following is required:
- Access to system server as user 'root'
- Networking setup and fully configured to port forward (or firewall opened) to ports 22, 80 and 443.

With above in place our technicians will be able to troubleshoot issues. We regret that we are not able to support systems, that do not satisfy above requirements. We ask for understanding.

If not too sure how to install PBXware we offer a professional installation service. Please contact sales or visit our web site for full details.

2.2 Installation Guide

The PBXware can be installed by one of the following installation methods:

CD ROM 7
CD ROM installation method is used to install the PBXware onto a commodity PC/server hardware. The installation process installs a Linux operating system, PBXware and all other necessary applications onto the system hard drive. Installation is easy, fast and includes everything needed to successfully install and operate.

WARNING:
CDROM will install the PBXware on hard disk and will erase all existing data, operating system and other files.

Virtual Machine 7
Often referred to as a "runtime environment". VMware virtual machine is a machine completely defined and implemented in software rather than hardware. This method is suitable for testing of up to 3 extensions since the virtual machine technology performance is limited. This edition is FREE and easy

to install as any other MS windows application.

Appliances 7
Ready to go, pre-configured appliances which do not need any installation sorted by the type of form factor and performance requirements.

Any of the above formats will boot the machine and all necessary software allowing administrator to login with browser into this machine. The Administrator can then license the system by entering the licence number. The system will contact our licensing server for authorization. Upon successful licensing, systems can be used normally by logging 37 into the system.

2.2.1 CD ROM

To install the system using the CD ROM do the following:

1. Download and burn CD image from http://downloads.bicomsystems.com/pbxware.iso
2. Boot up the system, wait for installer to start or you can press enter to start immediatelly
3. The system will be installed and rebooted . A boot message "Starting Installation" will show on reboot. This process may take few minutes. Please be patient. Once the installer has finished, the display will show obtained IP address.
4. Please login into setup wizard using your favorite web browser by navigating to http://$IPADDRESS (For Example: http://192.168.1.2). Default username is "root". Default password is "pbxware"

IMPORTANT:

- Before continuing please read system requirements. 5
- Setup wizard has security username/password in order to prevent unauthorized access. Defaults are: username: root, password: pbxware**.** Setup wizard will ask for the password to be changed in one of the setup wizard steps. After it is changed, It is very important to remember this password since it is system root and setup wizard password.

2.2.2 Virtual Machine

To install the system using the virtual servers do the following:

1. Order SOHO PBXware edition making sure to select "VMware Image" in the checkout as delivery method

2. After completing the PBXware order you will receive a confirmation email with download URL

3. Download the PBXware VMware image from
http://downloads.bicomsystems.com/vmware/pbxware-vm.zip and extract it.

4. Download 'VMware Player' from http://www.vmware.com/products/player/ location and install it on your computer

5. Navigate to the extracted PBXware VMware image folder and double clicking on the "PBXware" file

6. VMware player will boot up the virtual machine and display IP address

7. Point your browser to IP address displayed at VMware console screen. For example http://192.168.0.1

8. Click on the 'Factory Reset' which will be displayed on the screen. This process will take few minutes.

9. After factory reset is done, standard Setup Wizard login screen will show.

IMPORTANT

- Up to 3 (three) extensions supported
- Please make sure that system date and time zone are set correctly before executing the installation script
- Before continuing please read system requirements.
- Please make sure there is a valid DHCP server on your network

2.2.3 Appliances

In this chapter we will cover:

2.2.3.1 miniWALL

miniWALL delivers low noise perfomance for up to 32 extensions in a wall mount or desktop form factors.

Capacity

- 32 IP Extensions
- 8 Analog Extensions
- 8 Analog Lines
- 16 ISDN BRI Lines
- 30 ISDN PRI Lines
- 32 Concurrent Calls

Expansion Slots

- 2x Half Lenght 32bit/33MHz PCI Slots

Memory

- 512MB SDRAM Memory

Regulatory & Safety

- FCC, CE Regulatory & Safety Certification

Storage

- 128MB Compact Flash
- 40GB 2.5" Hard Disk Drive

Software Editions

- Business Edition
- Call Centre Edition

Form Factor
- Heavy Duty Steel Chassis
- Desktop/Wall Mount
- Industrial Grade Construction
- Low Noise

Processor(s)
- VIA C7 1.5GHz

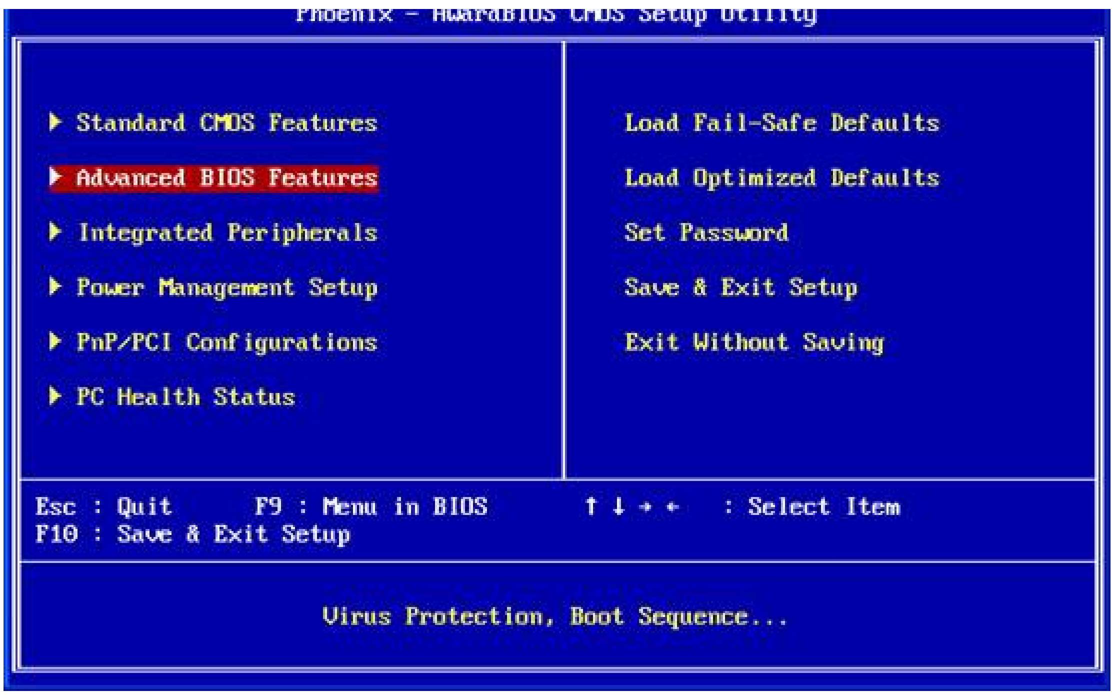
Operating System
- Gentoo Linux

NOTE: For serial connection settings please use '19200', '8', 'N', '1'

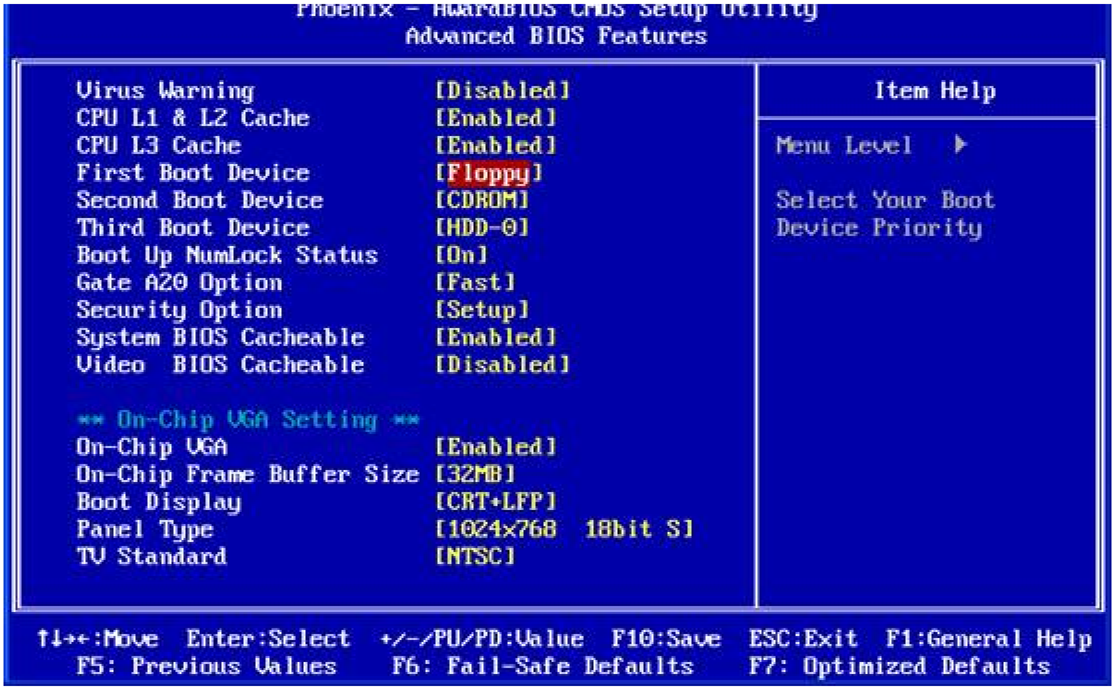
- How To Set Fist Boot Device From USB-CDROM 9

2.2.3.1.1 How To Set Fist Boot Device From USB-CDROM

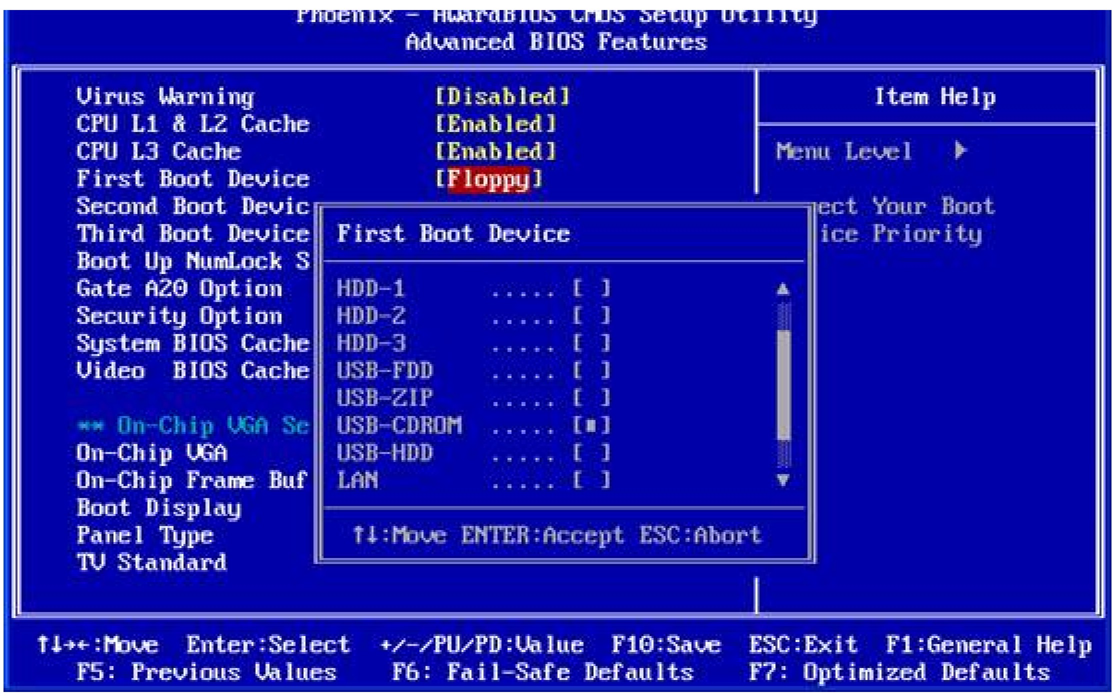
Step 1:



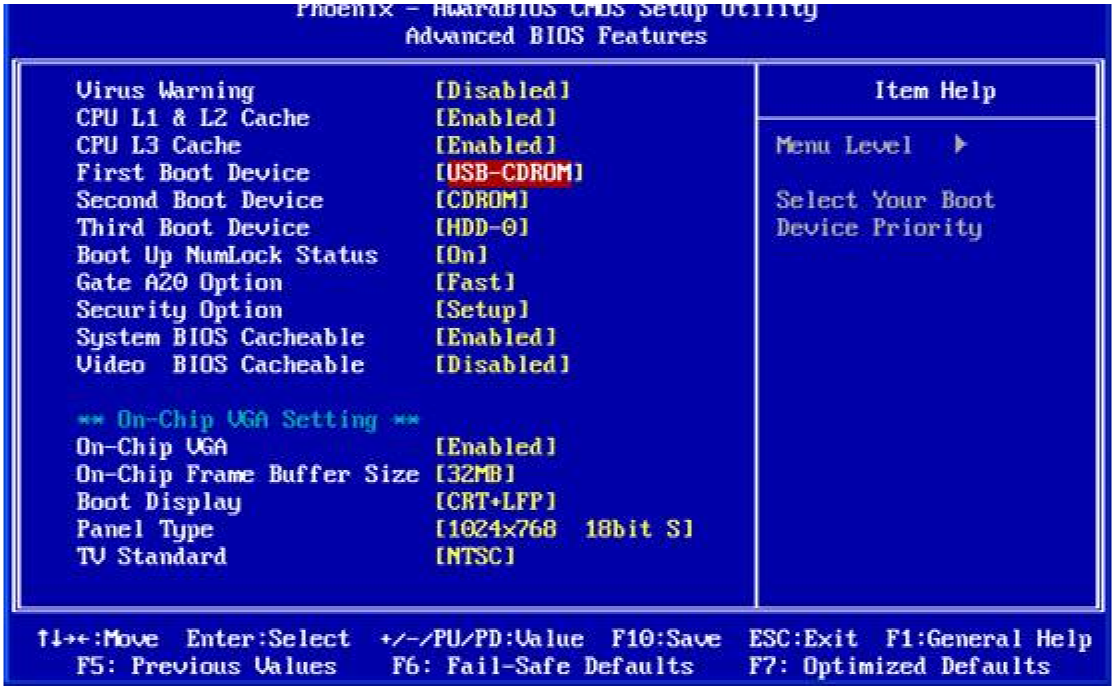
Step 2:


Advanced BIOS Features
Virus Warning [Disabled]
CPU L1 & L2 Cache [Enabled]
CPU L3 Cache [Enabled]
First Boot Device [Floppy]
Second Boot Device [CDROM]
Third Boot Device [HDD-0]
Boot Up NumLock Status [On]
Gate A20 Option [Fast]
Security Option [Setup]
System BIOS Cacheable [Enabled]
Video BIOS Cacheable [Disabled]
** On-Chip VGA Setting **
On-Chip VGA [Enabled]
On-Chip Frame Buffer Size [32MB]
Boot Display [CRT+LFP]
Panel Type [1024x768 18bit S]
TV Standard [NTSC]
Item Help
Menu Level
Select Your Boot Device Priority
↑↓→←:Move Enter:Select +/-/PU/PD:Value F10:Save ESC:Exit F1:General Help
F5: Previous Values F6: Fail-Safe Defaults F7: Optimized Defaults

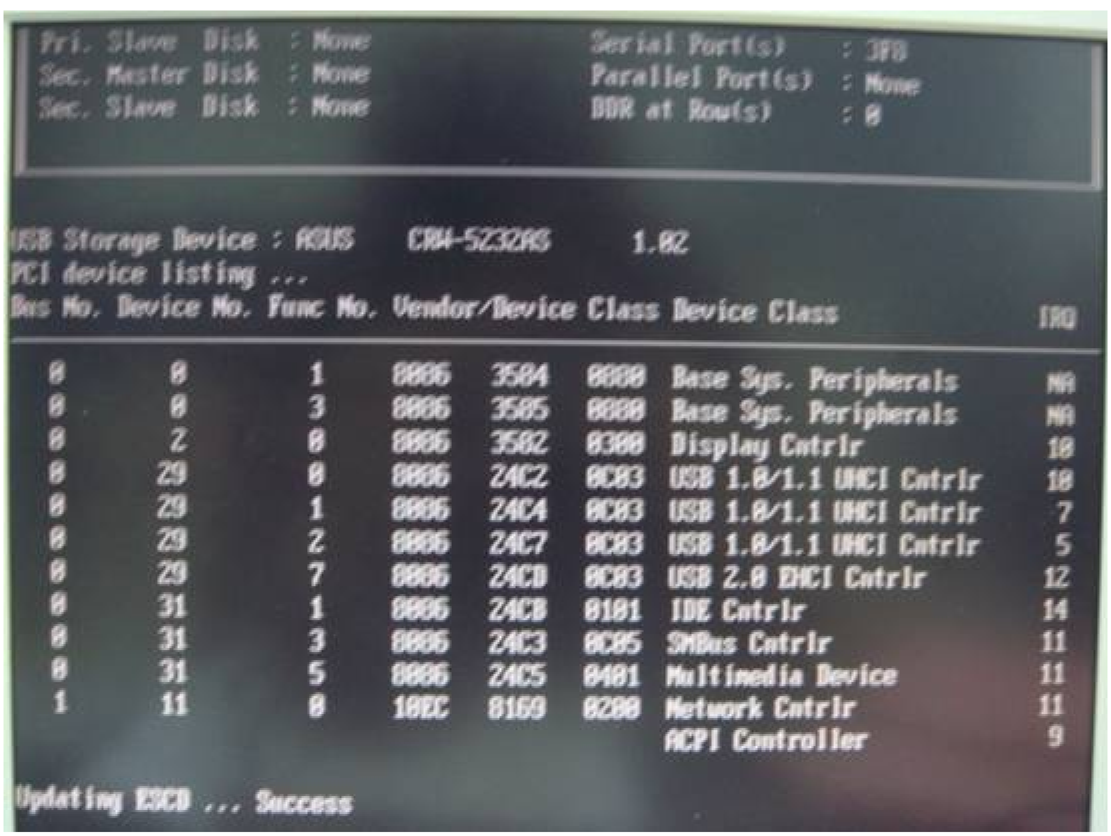
Step 3:


Advanced BIOS Features
Virus Warning [Disabled]
CPU L1 & L2 Cache [Enabled]
CPU L3 Cache [Enabled]
First Boot Device [Floppy]
Item Help
Menu Level
First Boot Device
HDD-1 ..... [ ]
HDD-2 ..... [ ]
HDD-3 ..... [ ]
USB-FDD ..... [ ]
USB-ZIP ..... [ ]
USB-CDROM ..... [■]
USB-HDD ..... [ ]
LAN ..... [ ]
↑↓:Move ENTER:Accept ESC:Abort
↑↓→←:Move Enter:Select +/-/PU/PD:Value F10:Save ESC:Exit F1:General Help
F5: Previous Values F6: Fail-Safe Defaults F7: Optimized Defaults

Step 4:



Step 5:
You will see the “USB Storage Device : ASUS CRW-5232AS 1.02” during boot process.



Step 6:

2.2.3.2 miniRACK

Capacity
- 32 IP Extensions
- 8 Analog Extensions
- 8 Analog Lines
- 32 ISDN BRI Lines
- 30 ISDN PRI Lines
- 32 Concurrent Calls

Expansion Slots
- 2x Half Lenght 32bit/33MHz PCI Slots

Memory
- 512MB SDRAM Memory

Regulatory & Safety
- FCC, CE Regulatory & Safety Certification

Storage
- 128MB Compact Flash
- 40GB 2.5" Hard Disk Drive

Form Factor
- 1 U Rackmount
- Heavy Duty Steel Chassis
- Industrial Grade Construction
- 282mm Rack Depth
- Low Noise

Processor(s)
- VIA C7 1.5GHz

2.2.3.3 maxiRACK

maxiRACK features up to 128 extensions with maximum flexibility in a rack mount format.

Capacity
- 256 IP Extensions
- 96 Analog Extensions
- 96 Analog Lines
- 64 ISDN BRI Lines
- 120 ISDN PRI Lines
- 128 Concurrent Calls

Expansion Slots
- 4x Full Lenght 32bit/33MHz PCI Slots

Memory
- 1-4GB DDR Memory

Regulatory & Safety
- FCC, CE Regulatory & Safety Certification

Storage
- 2x 250GB, RAID1 3.5" IDE Hard Disk Drive(s)

- 128MB Compact Flash

Software Editions
- Business Edition
- Call Centre Edition

Form Factor
- 2 U Rackmount
- Heavy Duty Steel Chassis
- Industrial Grade Construction
- 442mm Rack Depth

Processor(s)
- Intel Pentium 4 2.8GHz

Operating System
- Gentoo Linux

2.2.3.4 ftRACK

ftACK system combine proven fault-tolerant, pair-and-spare architecture in servers that deliver 99.999% uptime right out of the box.

Every ftRACK server uses replicated, fault-tolerant hardware to eliminate single points of failure and protect data integrity. Major components - CPUs, memory boards, input/output controllers, buses, power supplies, and fans - are duplexed and operate in lockstep. In the event of failure, the paired component continues normal operation.

This use technology eliminates the operational complexity and high costs inherent in high-availability alternatives. Hardware-based fault tolerance requires no failover scripting, repeated test procedures, or application modifications to ensure applications availability or smooth integration of systems into telecom environments.

ftRACK is widely deployed development and runtime software for applications in SS7, IP, and converged networks. Together these carrier-grade platforms bring you proven, affordable reliability, rapid time to market for new voice and data services, and lower development and operational costs.

Capacity
- 256 IP Extensions
- 128 Concurrent Calls

Expansion Slots
- 2x Full Lenght 64bit/33MHz PCI Slots

Memory
- 3GB DDR 200MHz ECC Memory

Regulatory & Safety
- FCC, CE Regulatory & Safety Certification

Storage
- External USB FDD Drive
- 24x Slimline EIDE 5.25" CDROM
- 2x 73GB Hotplug U160 SCSI Hard Disk(s)

Software Editions
- Enterprise Edition
- Business Edition
- Call Centre Edition

Form Factor
- Fault Tolerant Hardware
- 4 U Rackmount
- Heavy Duty Steel Chassis
- Industrial Grade Construction

Processor(s)
- 2x Intel® Xeon™ 2.4GHz

Operating System
- Fault Tolerant Linux

2.3 Setup Wizard

Setup Wizard is designed to collect essential data in order to get system up and running. After the setup wizard has finished the system should be fully licensed, fully operational and ready for use.

In order to login into the setup wizard please point browser to:
https://$IPADDRESS:81/ (For Example:**https://192.168.1.2:81/**)

In this chapter we will cover:

IMPORTANT:
Setup wizard has security username/password in order to prevent unauthorized access.
Defaults are: username: **root** , password: **pbxware .** Setup wizard will ask for the password to be changed in one of the setup steps. After it is changed, It is very important to remember this password since it is

system root and setup wizard password.

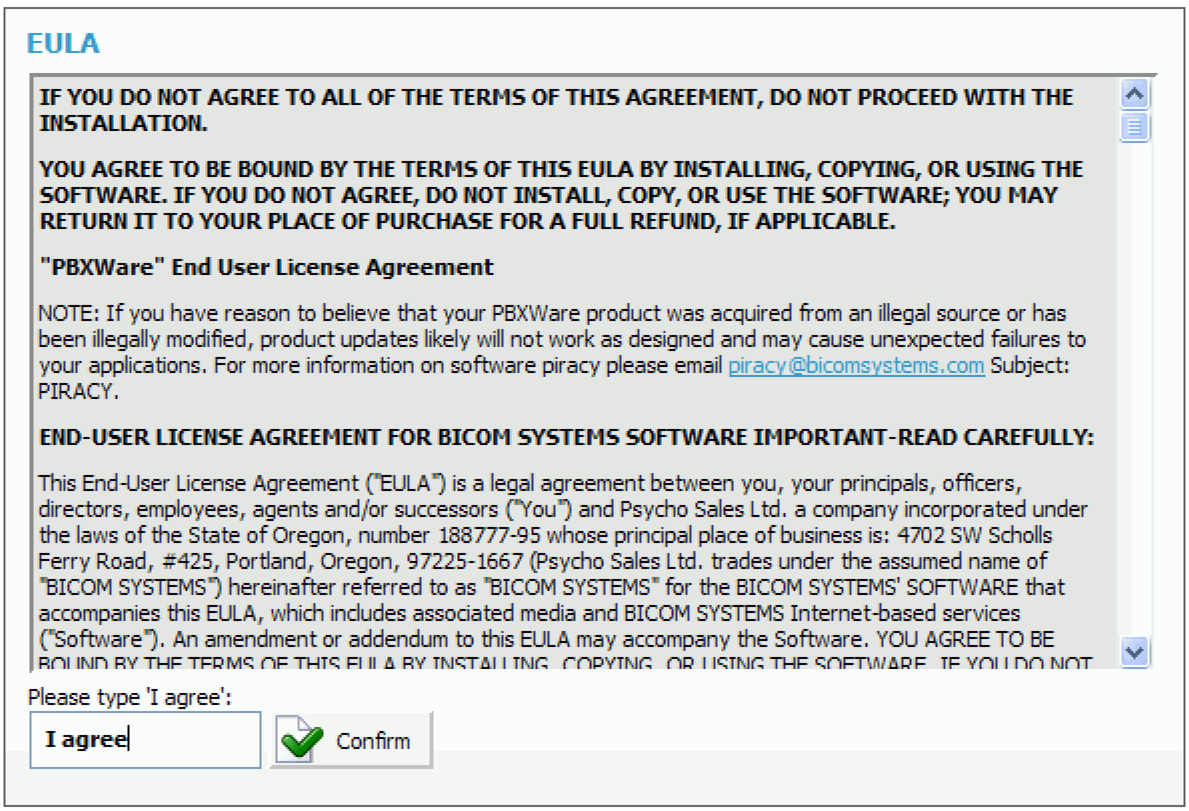
2.3.1 EULA

IMPORTANT:
Setup wizard has security username/password in order to prevent unauthorized access.
Defaults are: username: **root** , password: **pbxware .** Setup wizard will ask for the password to be changed in one of the setup steps. After it is changed, It is very important to remember this password since it is system root and setup wizard password.

EULA (end user license agreement) is the first setup wizard step. Please read the EULA and type "**I agree**" to proceed.

If not agreeing with EULA, please remove the installation media, remove system software and return license issued.

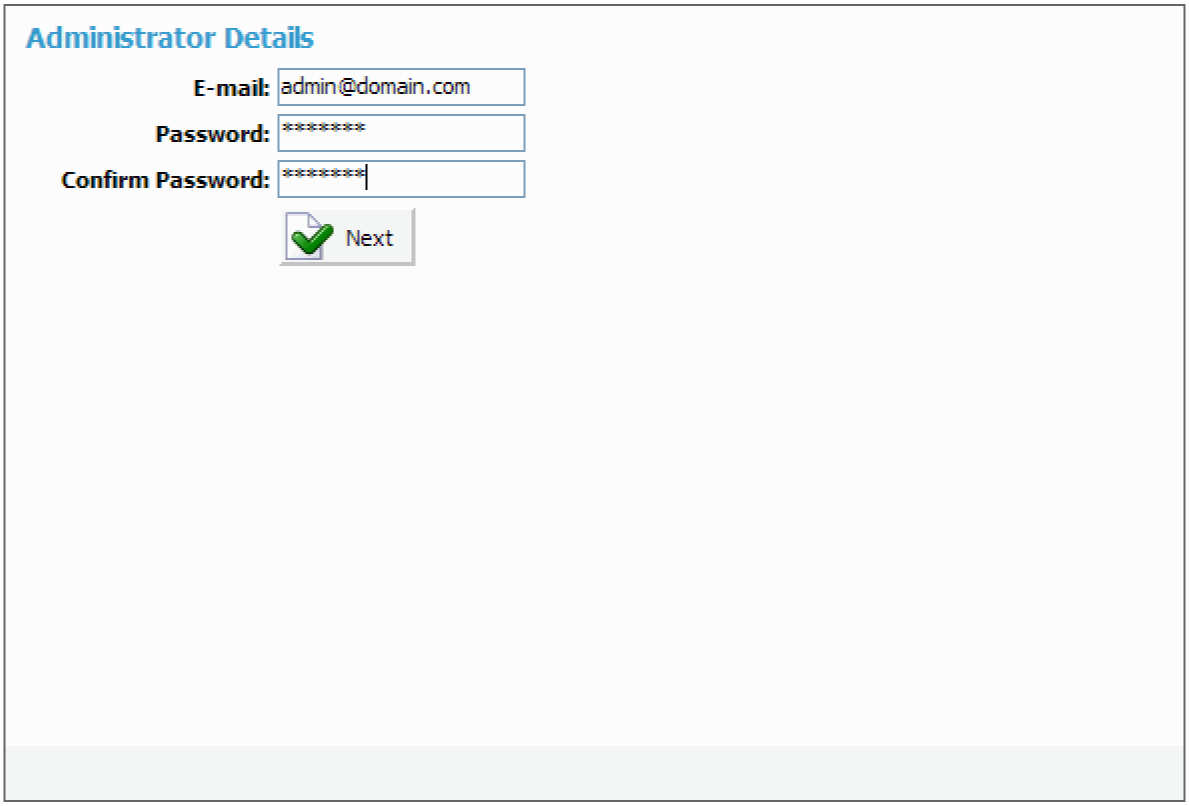
NOTE: This field is available for: Virtual Server, VMWare, CD, Appliance PBXware packages



2.3.2 Administrator Details

Provide details of user who will administer the system. These values are used when login into the PBXware

NOTE: This field is available for: Virtual Server, VMWare, CD, Appliance PBXware packages



Email:

Administrator email address

Example: Provided email address is used as a username for logging into system
Field Type: [a-z] [0-9] [@_.-]

Password/Confirm Password:

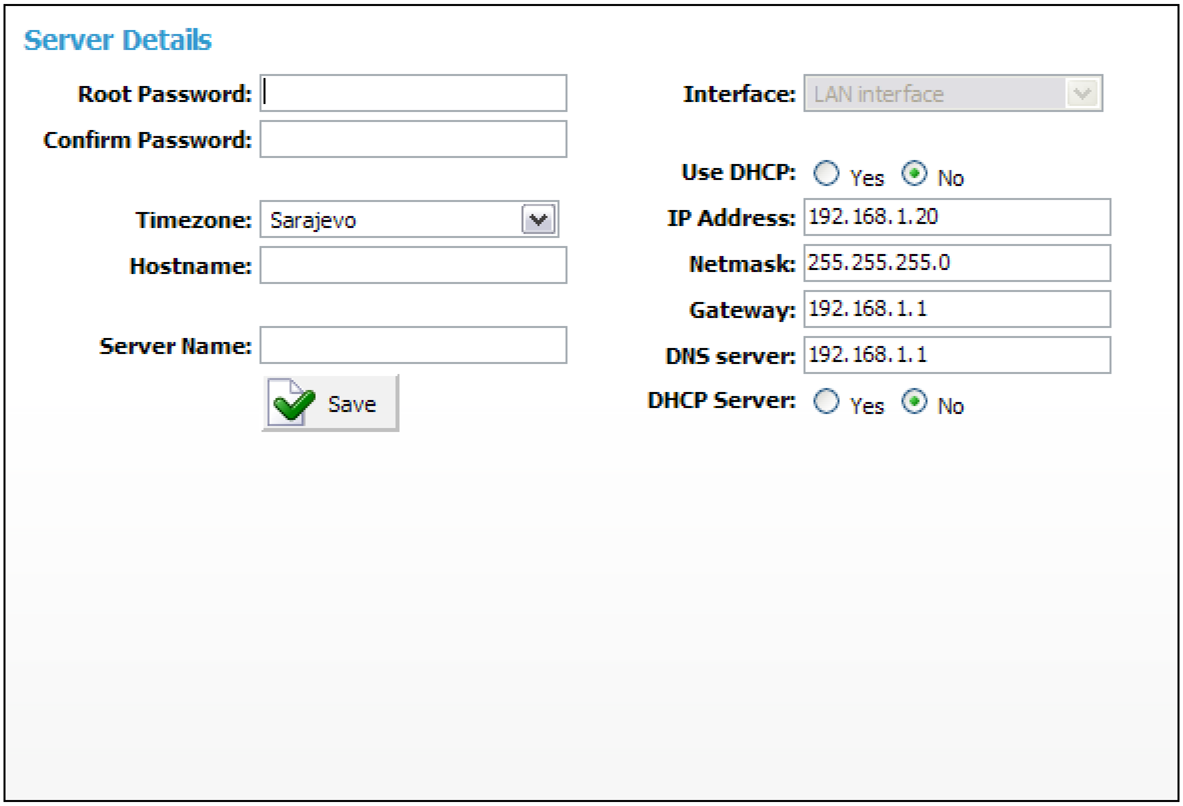
Administrator password

Example: Provided password is used for logging into system
Field Type: [a-z] [0-9]

2.3.3 System Details

These are system and network fields necessary for proper system operation.

NOTE: This field is available for: VMWare PBXware packages



Root Password:

Set the root password

Example: PBXware prompts for this password during the system/ssh login and when accessing system services throughinterface
Field Type: [a-z][0-9]

Confirm Password:

Confirm the root password

Example: Re-type the Root Password entered in the field above
Field Type: [a-z][0-9]

Time zone:

Time zone PBXware is located at

Example: Select the appropriate time zone, for example 'USA/East-coast
Field Type: Select box

Hostname:

The name given to machine which will identify the system on the network

Example: myhost
Field Type: [a-z][0-9]

Interface:

Interface PBXware uses (LAN/WAN)

Example: If the PBXware is in LAN interface, select it here. In some cases where PBXware is installed on appliances in WAN mode, select WAN here
Field Type: Select box

Use DHCP:

Is PBXware using DHCP or static IP address

Example: It is recommended to always set PBXware on static IP address (Set this option to 'No')
Field Type: Option buttons

IP Address:

Static PBXware IP address

Example: If PBXware is in the LAN provide its static IP address here
Field Type: [0-9]

Netmask:

IP address netmask

Example: This field is calculated automatically
Field Type: [0-9]

Gateway:

IF PBXware is located in LAN, set the gateway IP address here

Example: 192.168.1.1
Field Type: [0-9]

DNS server:

If PBXware is located in LAN, set the preferred DNS server IP address here

Example: 192.168.1.1
Field Type: [0-9]

DHCP Server:

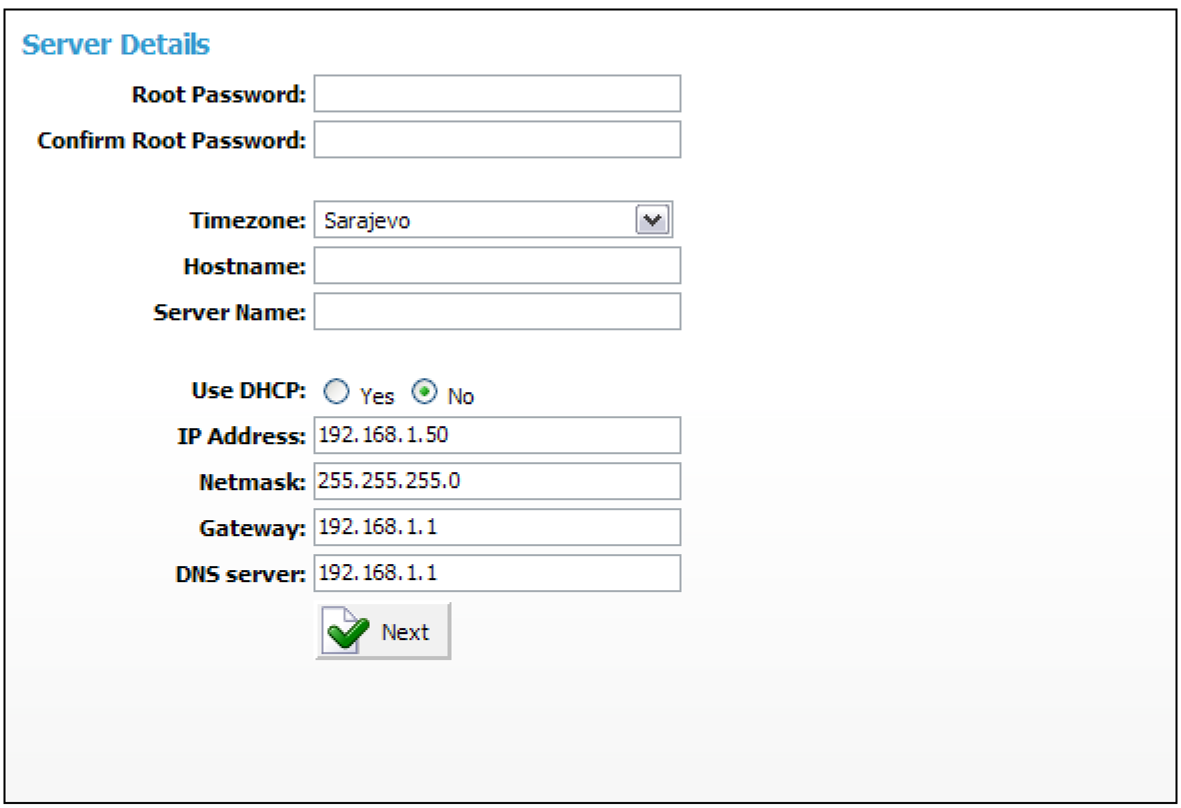
Should PBXware start its own DHCP server

Example: If there is no DHCP server on LAN, PBXware can start its own and provide UADs/Phones with this service
Field Type: [0-9]

2.3.4 Server Details

These are system and network fields necessary for proper system operation

NOTE: This field is available for: CD PBXware packages



Root Password:

Set the root password

Example: PBXware prompts for this password during the system/ssh login and when accessing system services throughinterface
Field Type: [a-z][0-9]

Confirm Password:

Confirm the root password

Example: Re-type the Root Password entered in the field above
Field Type: [a-z][0-9]

Time zone:

Time zone PBXware is located at

Example: Select the appropriate time zone, for example 'USA/East-coast
Field Type: Select box

Hostname:

The name given to machine which will identify the system on the network

Example: myhost
Field Type: [a-z][0-9]

Use DHCP:

Is PBXware using DHCP or static IP address

Example: It is recommended to always set PBXware on static IP address (Set this option to 'No')
Field Type: Option buttons

IP Address:

Static PBXware IP address

Example: If PBXware is in the LAN provide its static IP address here
Field Type: [0-9]

Netmask:

IP address netmask

Example: This field is calculated automatically
Field Type: [0-9]

Gateway:

IF PBXware is located in LAN, set the gateway IP address here

Example: 192.168.1.1
Field Type: [0-9]

DNS server:

If PBXware is located in LAN, set the preferred DNS server IP address here

Example: 192.168.1.1
Field Type: [0-9]

DHCP Server:

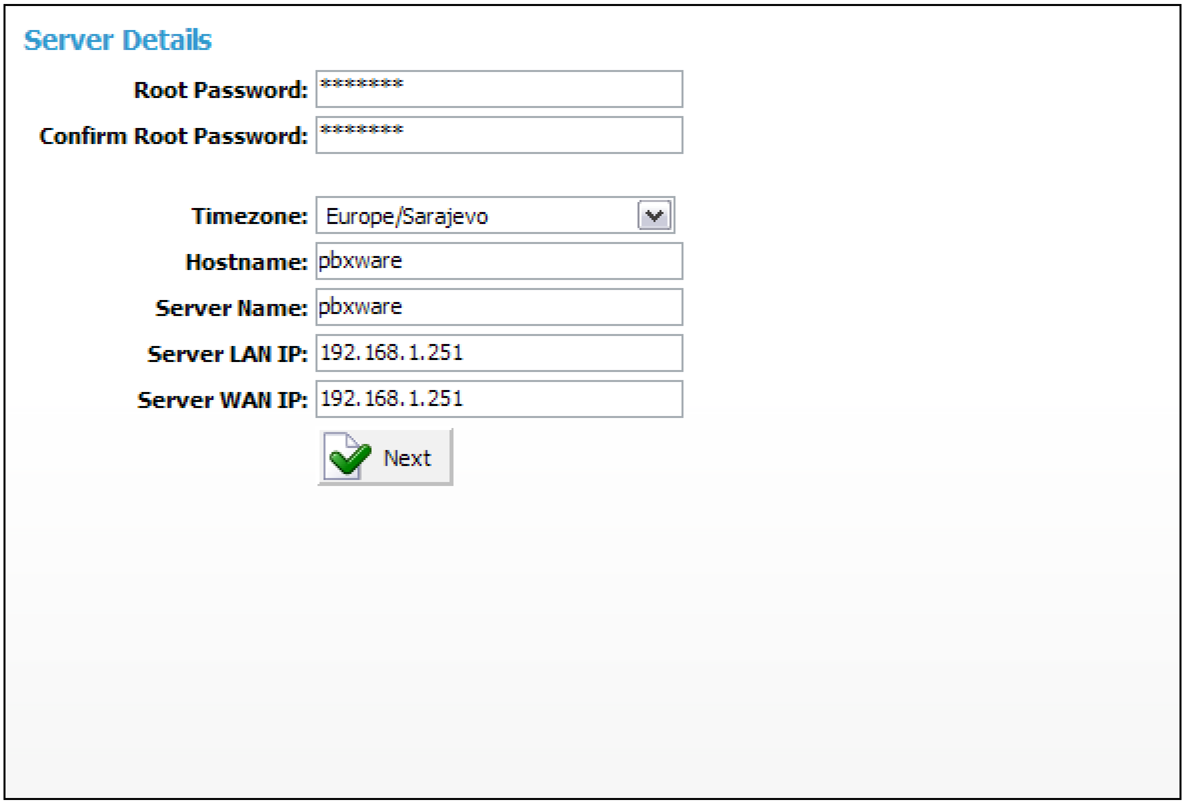
Should PBXware start its own DHCP server

Example: If there is no DHCP server on LAN, PBXware can start its own and provide UADs/Phones with this service
Field Type: [0-9]

2.3.5 Server Details

These are system and network fields necessary for proper system operation

NOTE: This field is available for: Virtual Server PBXware packages



Root Password:

Set the root password

Example: PBXware prompts for this password during the system/ssh login and when accessing system services throughinterface
Field Type: [a-z][0-9]

Confirm Password:

Confirm the root password

Example: Re-type the Root Password entered in the field above
Field Type: [a-z][0-9]

Time zone:

Time zone PBXware is located at

Example: Select the appropriate time zone, for example 'USA/East-coast
Field Type: Select box

Hostname:

The name given to machine which will identify the system on the network

Example: myhost
Field Type: [a-z][0-9]

Server name:

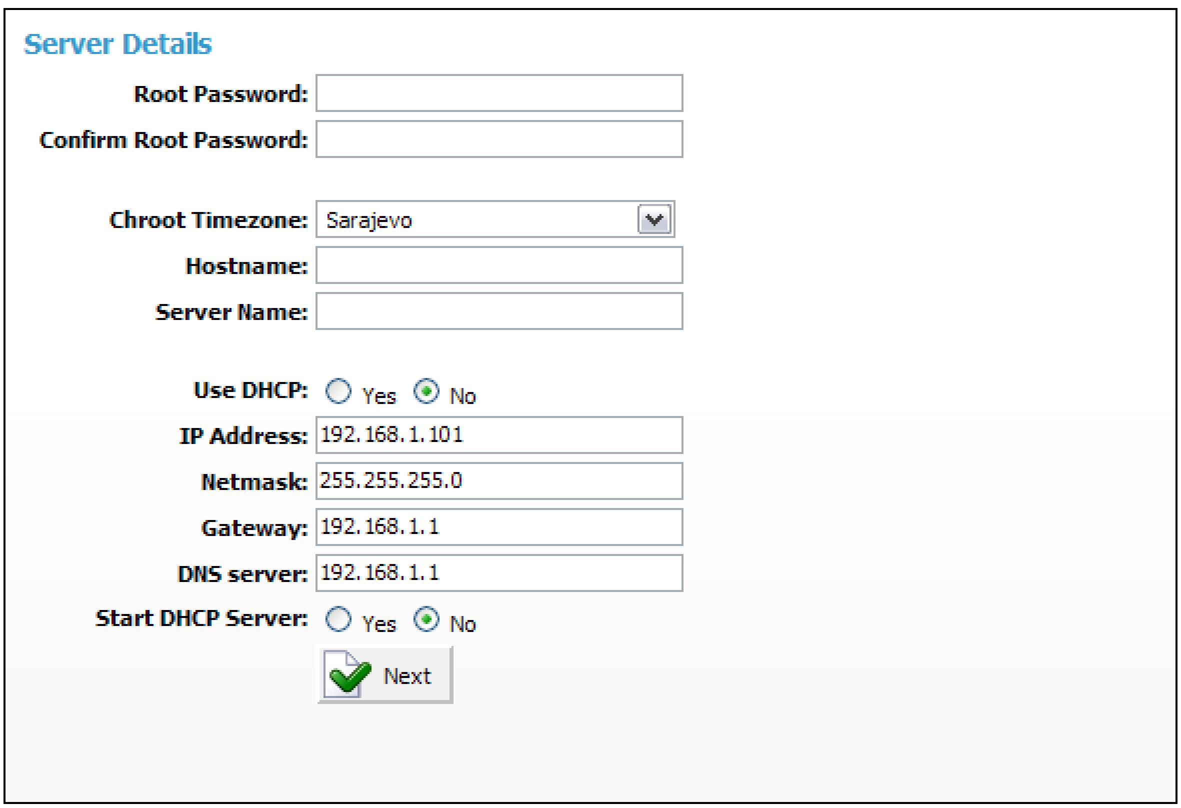
Virtually same as the 'Hostname' field, only this name will appear during system notifications, sent emails etc.

Example: domain.com
Field Type: [a-z][0-9]

2.3.6 Server Details

These are system and network fields necessary for proper system operation

NOTE: This field is available for: Appliance PBXware packages



Root Password:

Set the root password

Example: PBXware prompts for this password during the system/ssh login and when accessing system services throughinterface
Field Type: [a-z][0-9]

Confirm Password:

Confirm the root password

Example: Re-type the Root Password entered in the field above
Field Type: [a-z][0-9]

Chroot Time zone:

Time zone PBXware is located at

Example: Select the appropriate time zone, for example 'USA/East-coast
Field Type: Select box

Hostname:

The name given to machine which will identify the system on the network

Example: myhost
Field Type: [a-z][0-9]

Server name:

Virtually same as the 'Hostname' field, only this name will appear during system notifications, sent emails etc.

Example: domain.com
Field Type: [a-z][0-9]

Use DHCP:

Is PBXware using DHCP or static IP address

Example: It is recommended to always set PBXware on static IP address (Set this option to 'No')
Field Type: Option buttons

IP Address:

Static PBXware IP address

Example: If PBXware is in the LAN provide its static IP address here
Field Type: [0-9]

Netmask:

IP address netmask

Example: This field is calculated automatically
Field Type: [0-9]

Gateway:

IF PBXware is located in LAN, set the gateway IP address here

Example: 192.168.1.1
Field Type: [0-9]

DNS server:

If PBXware is located in LAN, set the preferred DNS server IP address here

Example: 192.168.1.1
Field Type: [0-9]

Start DHCP Server:

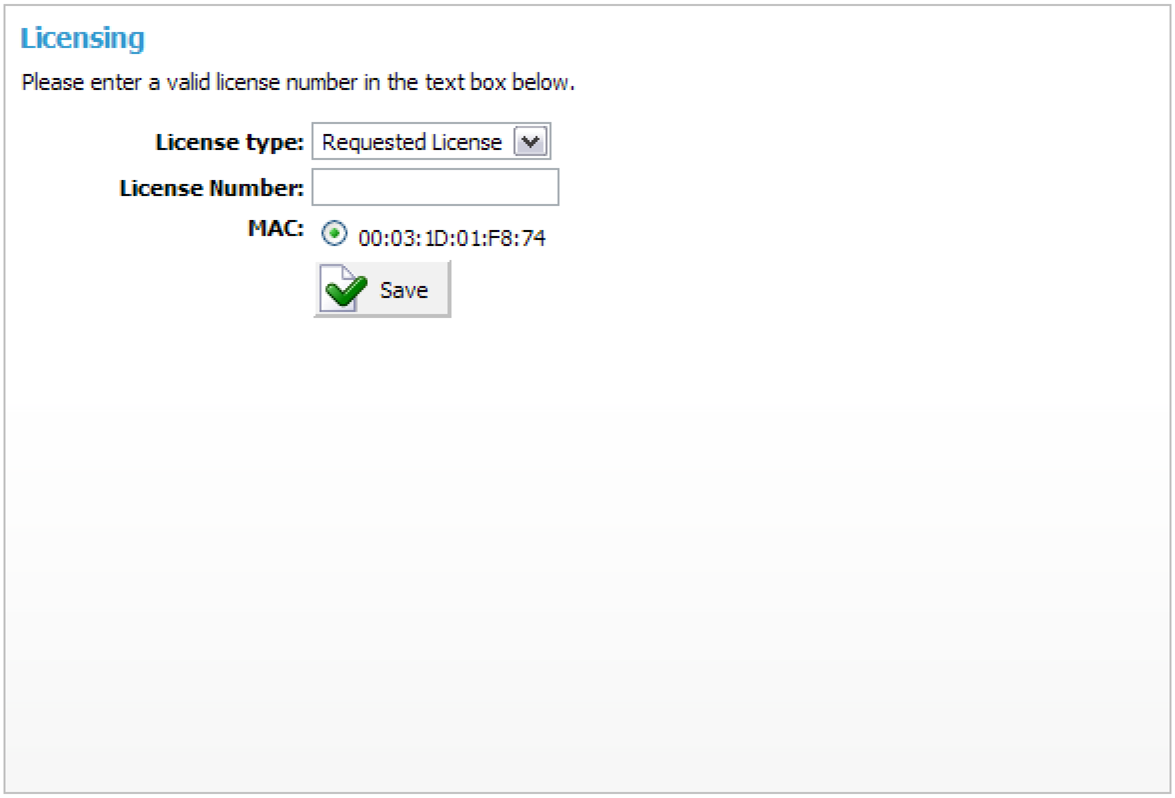
Should PBXware start its own DHCP server

Example: If there is no DHCP server on LAN, PBXware can start its own and provide UADs/Phones with this service
Field Type: [0-9]

2.3.7 Licensing

Licensing offers entering of a valid non active license and displays MAC address to which license will be valid with. If the system has more then one network adapter all found MACs will be shown for selection.

NOTE: This field is available for: Virtual Server, VMWare, CD, Appliance PBXware packages



License Number

PBXware license number

Example: Enter the PBXware license number, select the MAC address if more then one is present and click 'Next' to register the PBXware
Field Type: [a-z][0-9]

MAC:

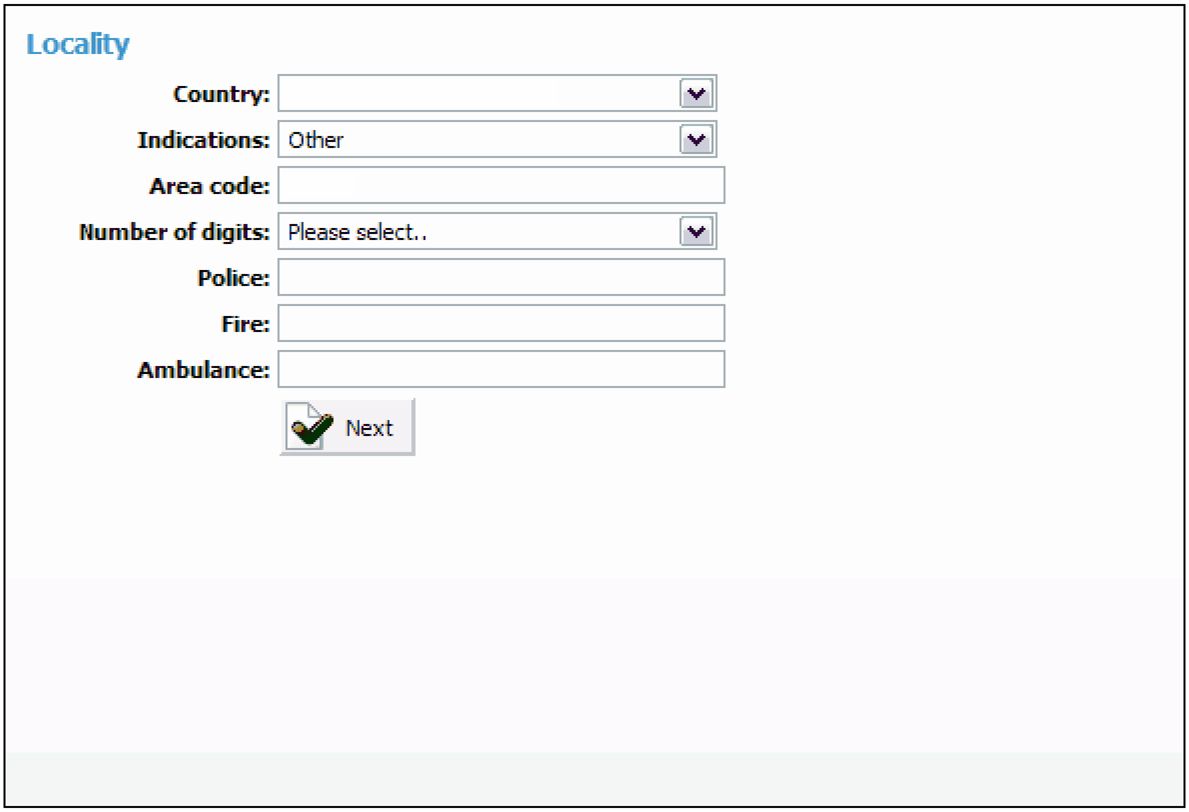
MAC address associated with the PBXware. NOTE: System must have access to fully operational Internet connection in order to license the system

Example: If more then one MAC address is present, select the one you whish to associate to PBXware and click 'Next'
Field Type: Select box

2.3.8 Locality

Locality allows for system 'local' values to be entered in order to setup all the necessary values for normal system operation

NOTE: This field is available for: Virtual Server, VMWare, CD, Appliance PBXware packages



Country:

Select the country PBXware is located at

Example: United Kingodm
Field Type: Select box

Indications:

Typical telephony sounds PBXware will use

Example: Different signal is heard when handset is picked up in different countries. This field will be set automatically. If your country is not present select 'Other'
Field Type: Select box

Area Code:

Area code of the city where PBXware is located

Example: If this PBXware is in New York, set '212' here
Field Type: [0-9]

Number Of Digits:

Number of digits PBXware will associate with local extensions

Example: If this field is set to '4', all local extensions will have a range from 1000-9999
Field Type: [0-9]

Police/Fire/Ambulance:

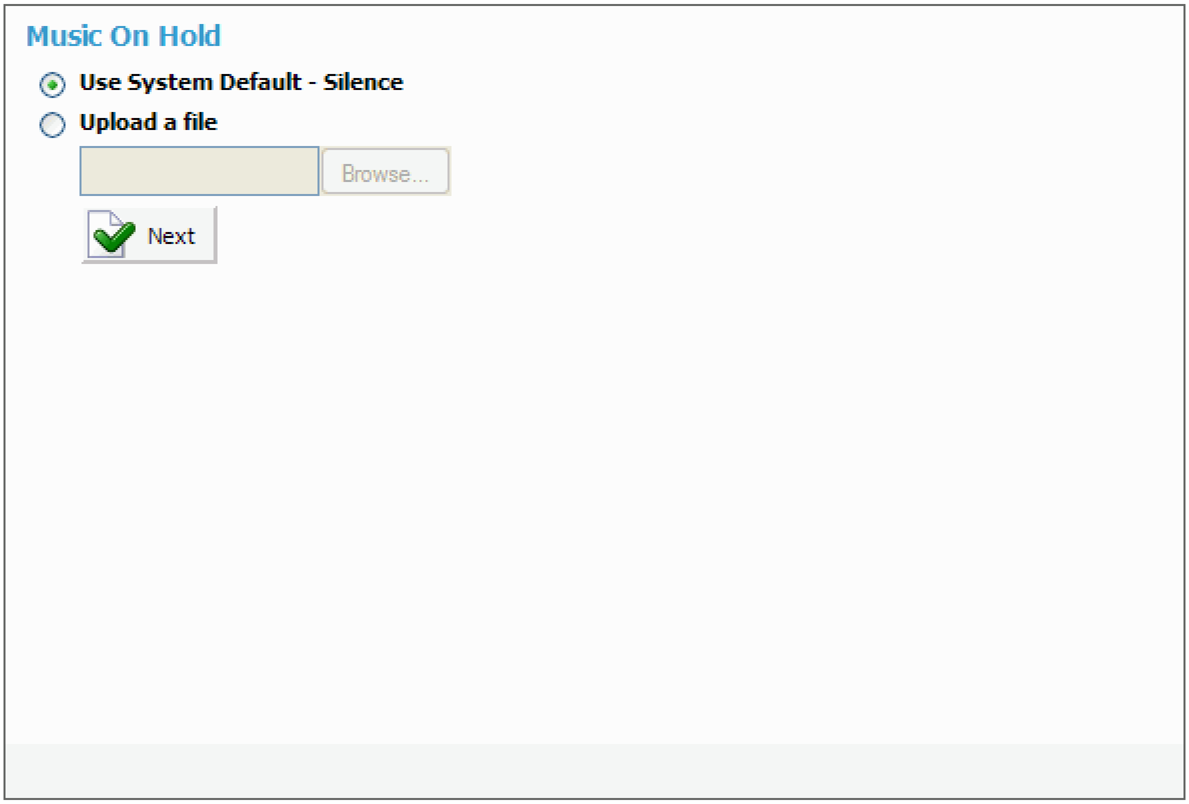
Number of Emergency Services in area where PBXware is located

Example: If in USA for example, set these fields to '911'. PBXware has an option to dial these emergency services through certain trunks. If all trunk channels are busy, an active call will get dropped in order to dial these
Field Type: Select box

2.3.9 Music On Hold

System comes with default 'silence' music on hold sound file or custom sound file can be uploaded. This file must be in .wav, gsm or mp3 format and not bigger then 8MB.

NOTE: This field is available for: Virtual Server, VMWare, CD, Appliance PBXware packages



Upload a file

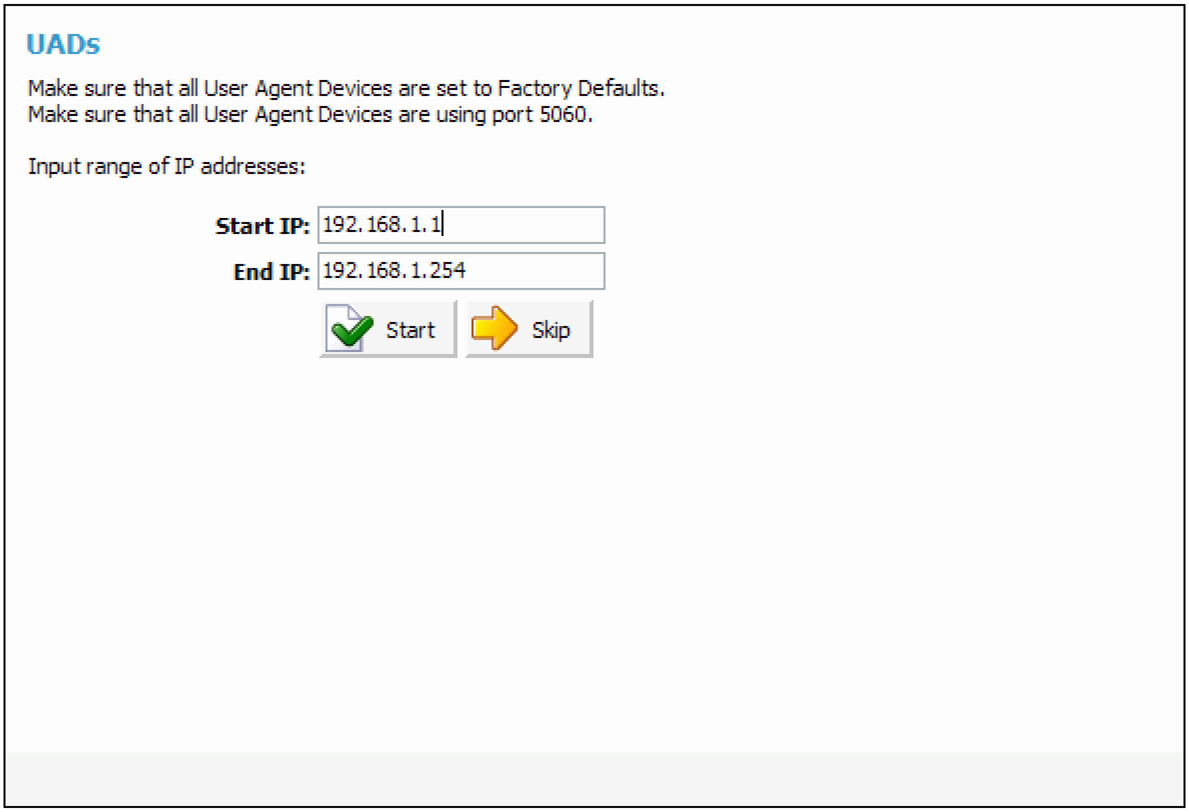
Upload a custom sound file from Desktop to PBXware server. NOTE: If you select 'Use System Default - Silence' no sound file will be uploaded and only 'Silence' file will exist in the 'System: MOH'

Example: Select this option and click the 'Browse' button to select a file on your Desktop and click 'Next' button to upload it on the PBXware.
Field Type: Select box

2.3.10 UAD

UAD step allows an installer to search the network for supported devices. PBXware will automatically scan the network, list all devices found and offer auto configuration

NOTE:
Please make sure that all User Agent Devices are reset to Factory Defaults and using port 5060 before continuing.
This field is available for: Virtual Server, VMWare, CD, Appliance PBXware packages



Start IP:

Start the search from this IP address

Example: 192.168.1.1
Field Type: [0-9]

End IP:

End the search with this IP address

Example: 192.168.1.254
Field Type: [0-9]

Start:

Start UAD/Phone network search

Example: Enter the Start and End IP address and click this button to perform the search
Field Type: Command Button

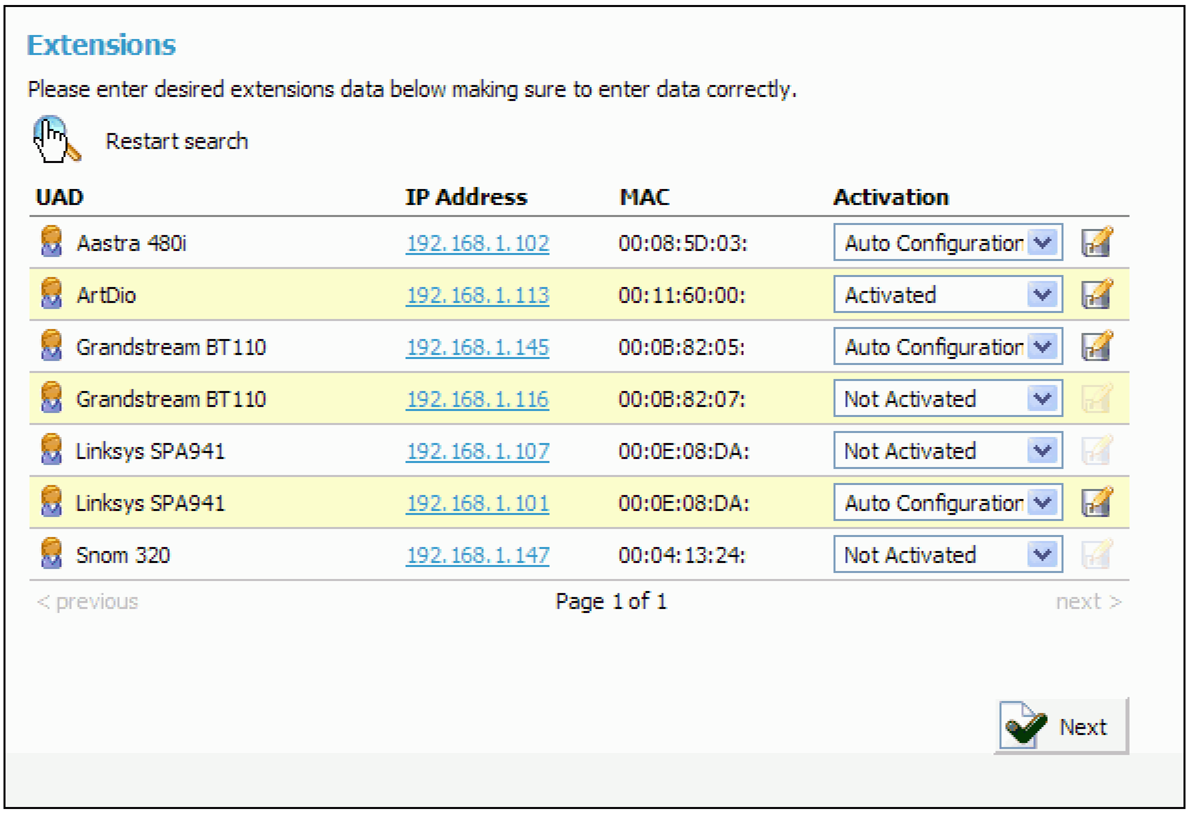
Skip:

Skip this step

Example: Skip the search for UADs/Phones on local network and proceed to next Setup Wizard step
Field Type: Command Button

2.3.10.1 Extensions

After selecting 'default' UAD this step offers extension fields in order to add multiple users extensions.



UAD:

Name of the UAD/Phone found in the local network

Example: Aastra 480i
Field Type: Display

IP Address:

IP address of the UAD/Phone found in the local network

Example: If the phone is located on the 192.168.1.102 IP address that address will be displayed here. Click it to open the UAD/Phone settings in new browser window
Field Type: Display

MAC:

Mac address of the UAD/Phone found in the local network

Example: 00:0B:27:22:08
Field Type: Display

Activation:

Set the preferred activation method. NOTE: If UAD/Phone supports the 'Auto Configuration' option

Example:
- **Not Activated**: Skip/Do not configure this UAD/Phone
- **Activated**: Creates the PBXware extension and nothing else
- **Auto Provisioning:** Creates PBXware extension and a file in TFTP directory for provisioning. This step requires manual reboot of a device
- **Auto Configuration:** Creates PBXware extension and a TFTP provisioning file. Goes into web interface of the phone, automatically configures all details and reboot the device'

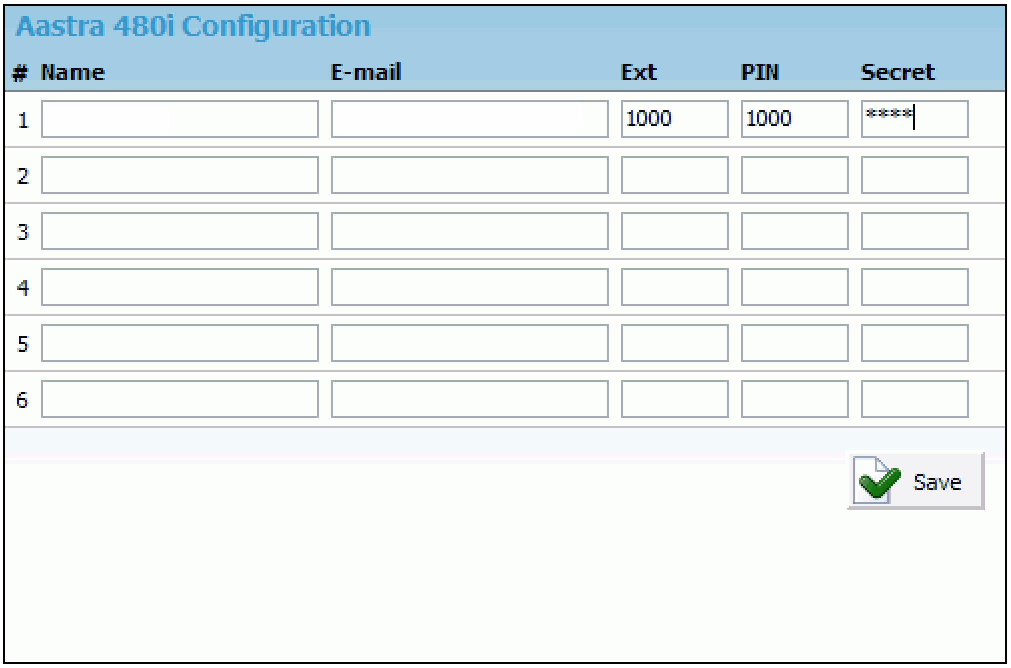
Field Type: Display


Edit the UAD/Phone configuration
Example: Click to edit UAD configuration
Field Type: Button

2.3.10.1.1 Edit

Add system extension for all UAD lines from this location by filling in the following fields:

NOTE: Number of extensions may vary from one phone to another



Name:

Name of the user associated with the extension. This name will be used as a Caller ID information when calling

Example: John Smith
Field Type: [a-z][0-9]

E-mail:

Email address associated with the extension. This email is used for system notification and logging into the Self Care

Example: john@domain.com
Field Type: [a-z][0-9]

Ext:

Network number

Example: 1001
Field Type: [0-9]

PIN (Personal Identification Number):

Password used for accessing voicemail and other additional PBXware services.

Example: When B wants to access his 'Voicemail', he is asked to authenticate with personal 4(four) digit PIN.
Field Type: [0-9]

Secret:

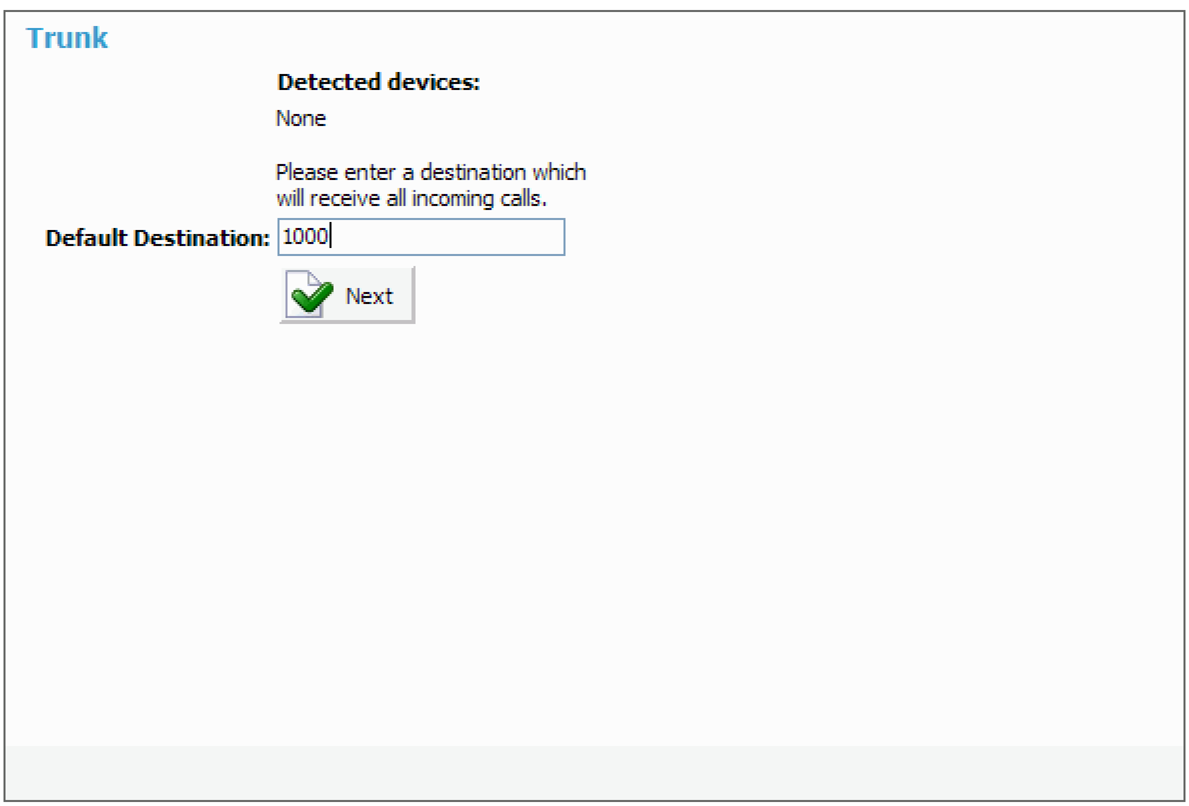
Password used for the UAD/Phone authentication with the PBXware.

Example: This fields is auto-generated and consists of letters and digits, but you can change this values
Field Type: [a-z] [0-9]

2.3.11 Trunk

Trunks step will try to detect supported trunk devices present on the system. Once detected, wizard will automatically create a trunk
based on most common configuration values. Please visit www.bicomsystems.com/docs/uad/ for a list of currently suppored devices.

NOTE: This field is available for: Virtual Server, VMWare, CD, Appliance PBXware packages



Detected devices:

If setup wizard detects any hardware devices (cards) they will be listed here

Example: -
Field Type: Display

Default Destination:

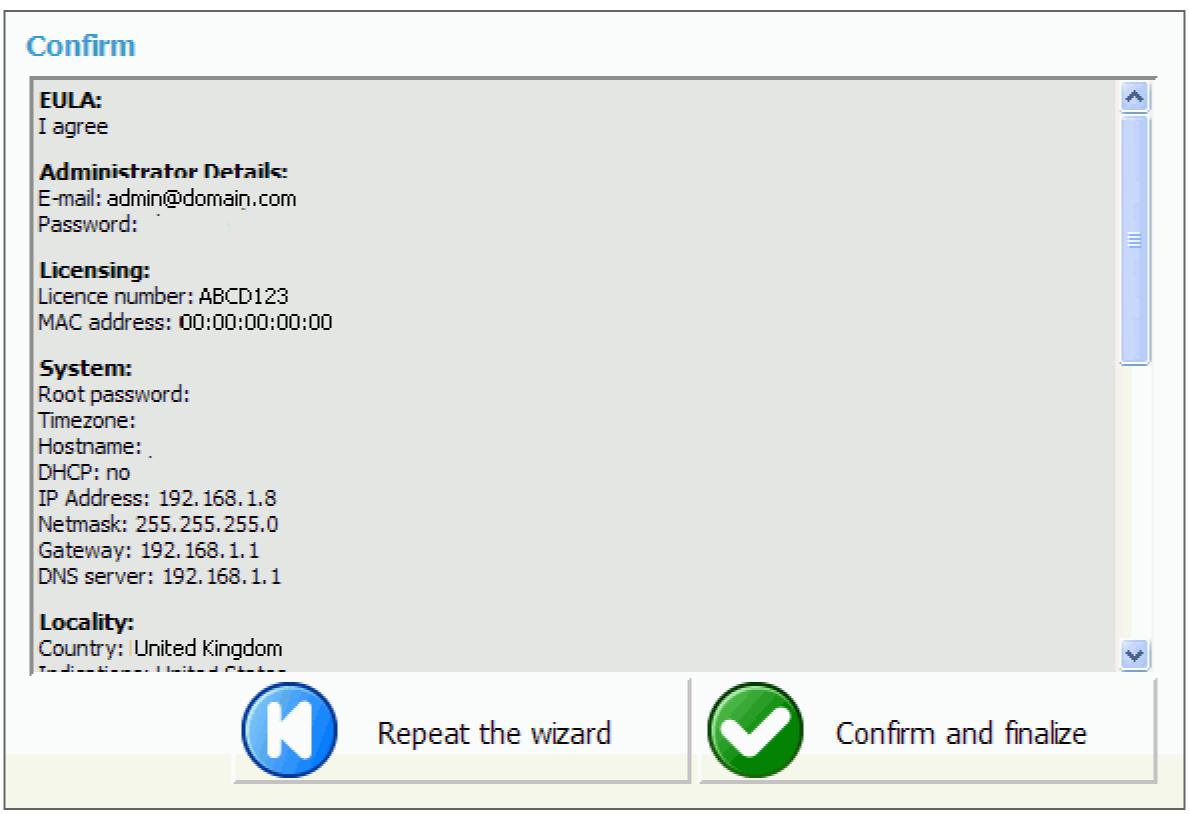
If PBXware has no DIDs set, all incoming call will go to this destination

Example: Set '1000' for example to make all calls coming from trunks to go to this extension
Field Type: [0-9]

2.3.12 Confirmation

Finally, the confirmation step allows for all values to be revised and to either finish the wizard or start all over from the beginning. If 'Confirm and finalize' is clicked on, setup wizard will finish and browser will be redirected to system login screen.

NOTE This field is available for: Virtual Server, VMWare, CD, Appliance PBXware packages



Repeat the wizard

Click this button to repeat the wizard

Example: All provided details will be reset and started again from the step one
Field Type: Command button

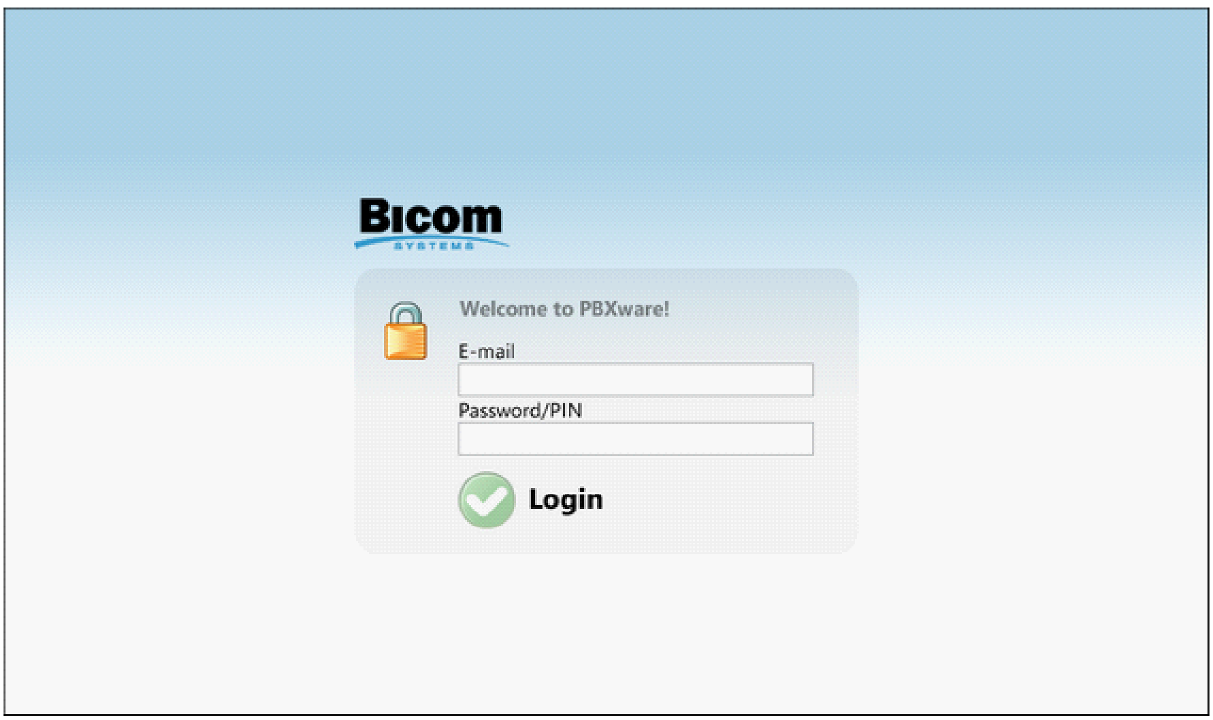
Confirm and finalize

Confirm the date provided in the setup wizard and finalize the installation

Example: All provided details will be applied and setup wizard will redirect to PBXware login screen
Field Type: Command button

2.4 Loging into the system

In order to login into the system please point browser to:
http://$IPADDRESS/ (For Example:**http://192.168.1.2/**)



Email:

Administrator email address

Example: This email address is set in the initial setup wizard
Field Type: [a-z] [0-9] [@_.-]

Password / PIN:

Administrator password

Example: This password is set in the initial setup wizard
Field Type: [a-z] [0-9]

2.5 Updates/Upgrades

In this chapter we will cover:
- Updates 38
- Upgrades 38
- License Upgrades 39

2.5.1 Updates

In order to update PBXware to a specific firmware version follow these steps:

1. Login into PBXware web interface

2. Navigate to 'Site Settings: Updates' and click on 'Updates' button

3. Enter update username=**'pbxware'** and password=**'update'** and wait until the system shows the interface again

2.5.2 Upgrades

PBXware uses even numbers for stable and odd numbers for development releases. Current stable release is 2.0 and development version is 2.1. To upgrade PBXware to 2.0 version for example do the following:

1. Login to PBXware via SSH

2. Stop all servers
/home/servers/sitemanager/sh/stop
/home/servers/pbxware/sh/stop
/home/servers/httpd/sh/stop

3. Modify:
'/home/servers/sitemanager/sh/update' by setting the PROJECT line to 'PROJECT=sitemanager-2.0'
'/home/servers/pbxware/sh/update' by settings the PROJECT line to 'PROJECT=pbxware-2.0'

'/home/servers/httpd/sh/update' by setting the PROJECT line to 'PROJECT=httpd-2.0'

4. Upgrade the software:
   /home/servers/sitemanager/sh/update
   /home/servers/pbxware/sh/update
   /home/servers/httpd/sh/update

NOTE: default update username: **pbxware**, password: **update**

5. Start all servers
   /home/servers/sitemanager/sh/start
   /home/servers/pbxware/sh/start
   /home/servers/httpd/sh/start

6. Navigate to PBXware interface and login.

7. Re-save at least one 'SIP' and IAX' extension, 'MOH(Music on Hold)' and a 'Queue', but going through all options and re-saving them is recommended

2.5.3 License Upgrades

To upgrade PBXware license do as follows:

1. Login into PBXware web interface

2. Navigate to 'Site Settings: Upgrades'

3. Enter the system root username and password (default ones 'pbxware'/'update')

4. Enter the license number in 'License Number' field (e.g. 0A9DS8F7)

5. Click on 'Save' button

Part III

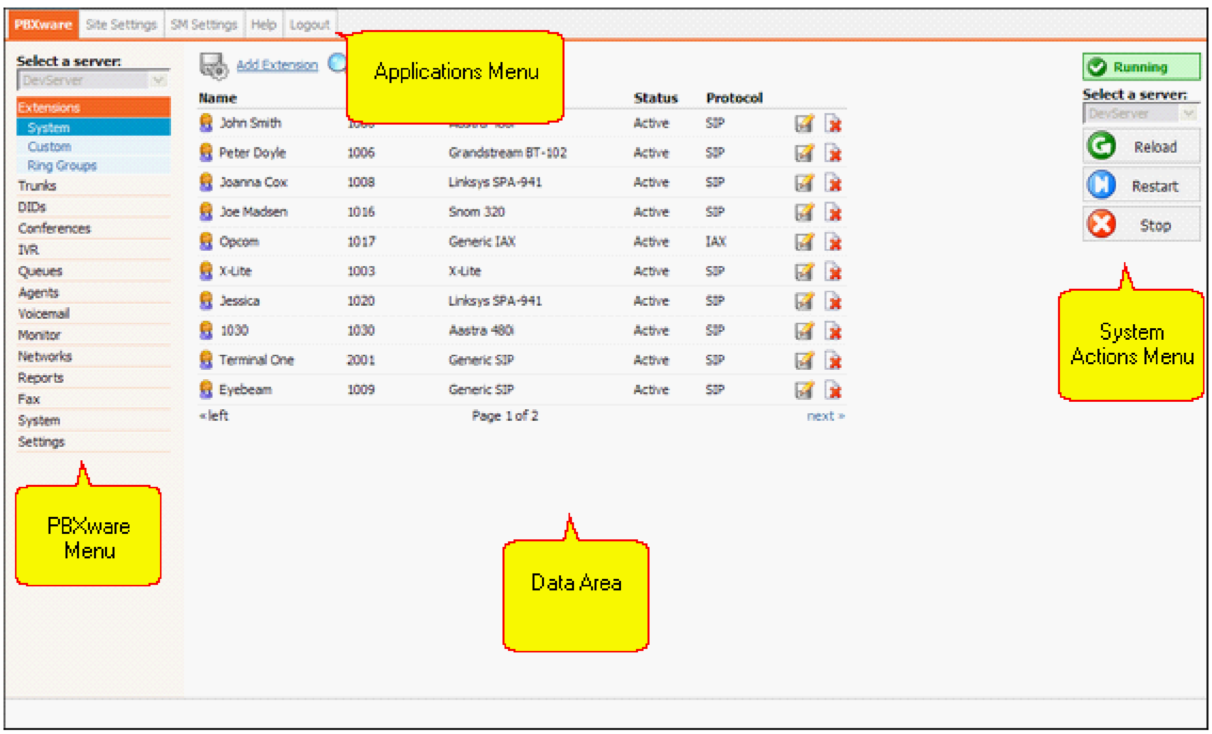
3 System Overview

In this chapter we will cover:

- Administration Interface 41
- Role Based Administration 42
- Standard and Advanced Options 43

3.1 Administration Interface

Administration interface consist of applications, administration and actions menus with all data displayed in data area.



Applications Menu:
Applications menu provides access to all applications, site settings, SM settings, log out and help links.

PBXware:
Clicking on PBXware will display its menu showing all available sections.

Site Settings:
This link allows managing of system users, groups, backup, sessions, updates and licensing.

SM Settings:
Options of system administrators management and currently available sites can be found here.

Help:
A link to help documentation.

Logout:
This link allows user to log out.

PBXware menu:
PBXware menu is located on the left and it displays all administration sections available.

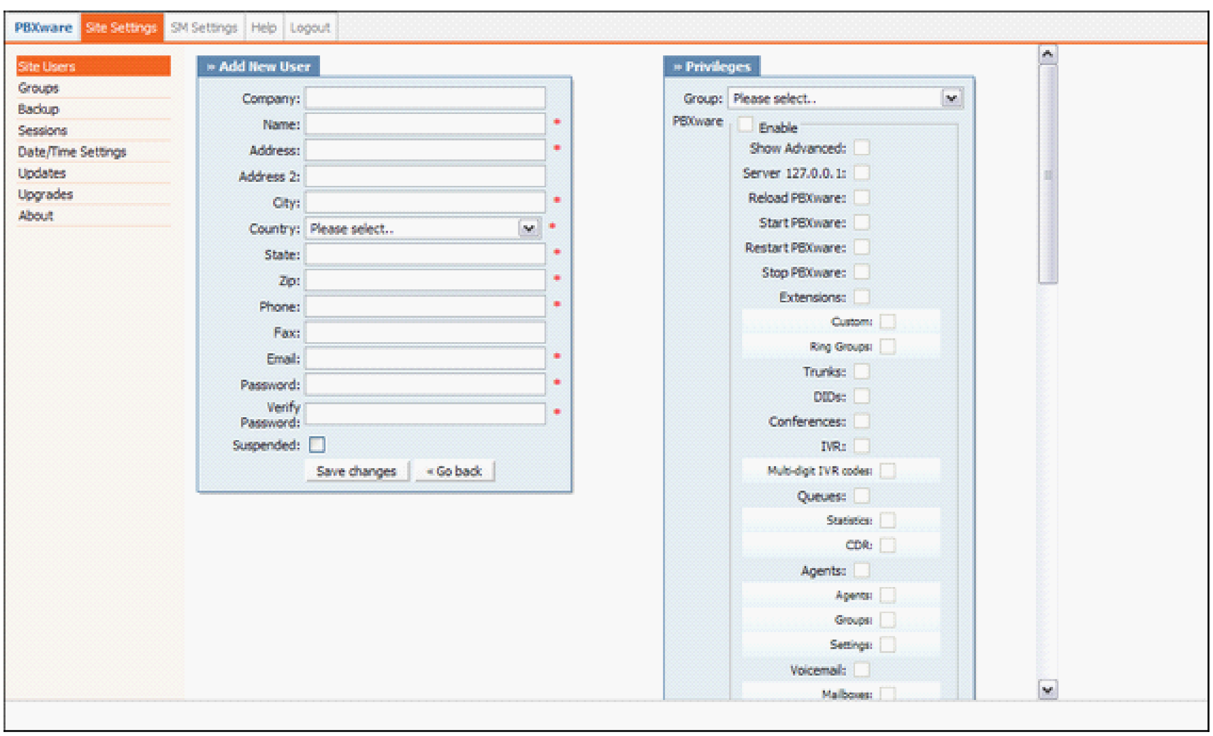
Data Area:
Data are displays all results from various actions performed.

System Actions Menu:
System actions status on the top right allows viewing of the system status and performing of start, stop and restart actions on one or more servers.

3.2 Role Based Administration

System administration can be delegated to various users in order to perform role based administration. An unlimited number of users and groups can be created by system administrator(s). Each user will only then have access to the sections of PBXware menus accordingly to the group membership permissions.

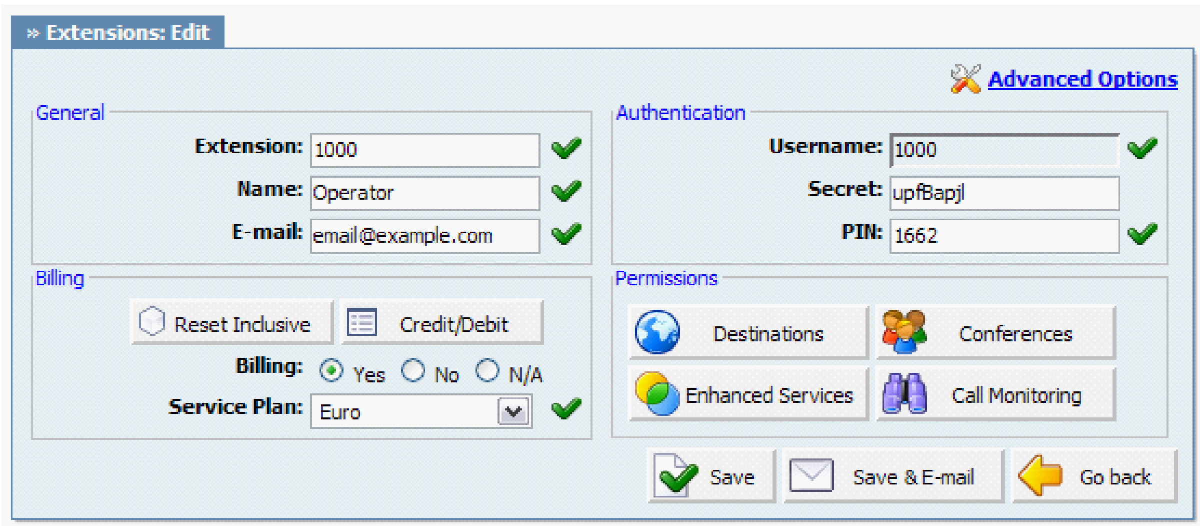
This is commonly used to allow management, operators, supervisors etc access to the sections of the PBXware to which they have adequate knowledge and experience.



3.3 Standard and Advanced Options

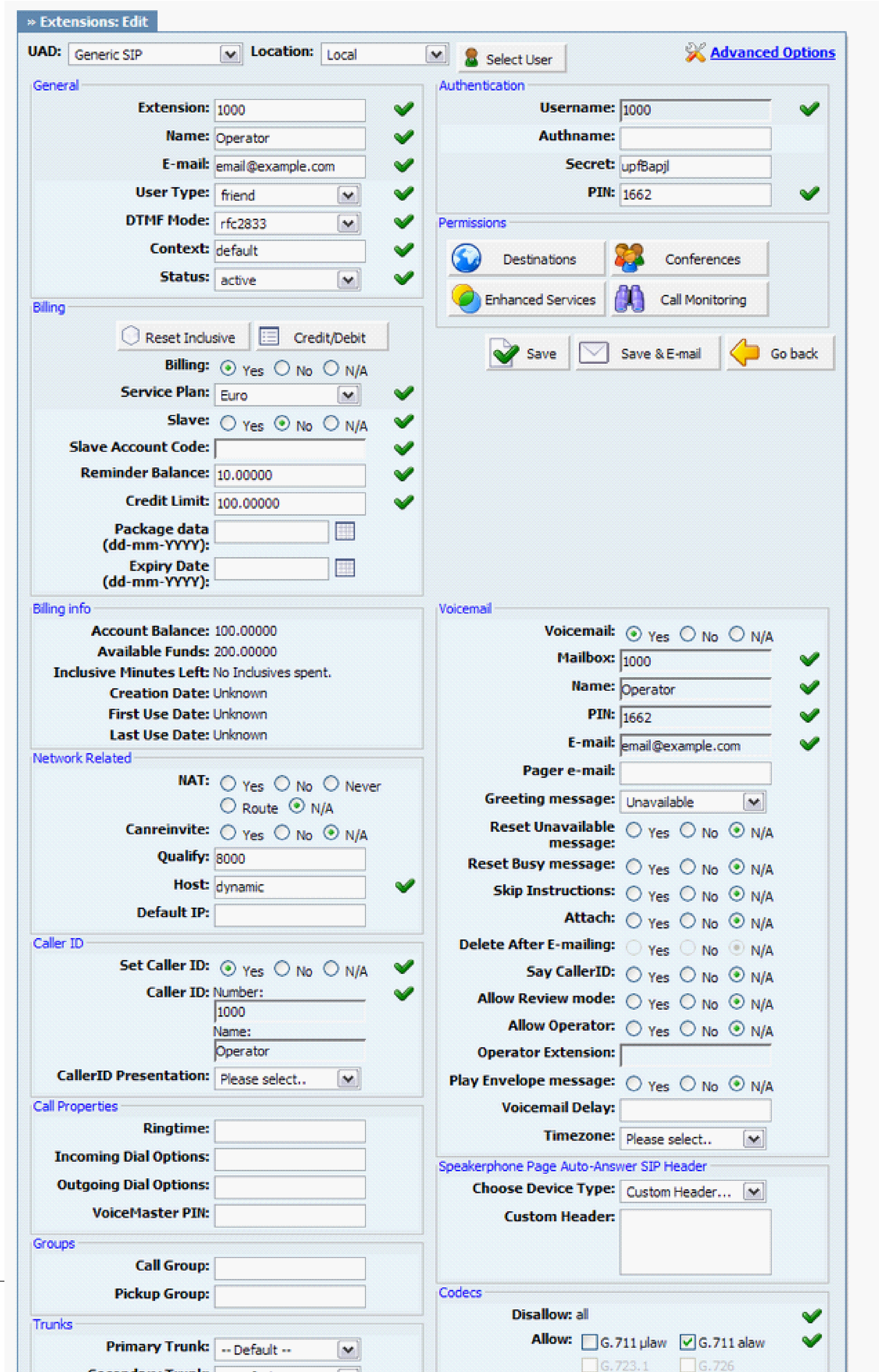
PBXware has been designed with simplicity and extensive configuration options as primary goals. In order to achieve both goals administration has standard and advanced modes throughout all sections.

Standard Options
Standard mode is designed to allow an easy 'no brainier' method of administration as the level of configuration knowledge needed is based on common fields of information (Name, email address etc.). This is possible to achieve since PBXware uses a powerful template system which pre-configures all advanced options leaving only common information values to be entered.



Sample of "Standard Options"

Advanced Options
Advanced mode options on the other hand requires much better system knowledge in order to fine tune the system settings for various applications and usages. This manual tries to provide detailed information for advanced options. However, administrators should bear in mind that extensive training and hands on experience is required in order to be able to administer advanced sections effectively. For simplicity, all options available in the standard options are not repeated in this manual under the advanced options.

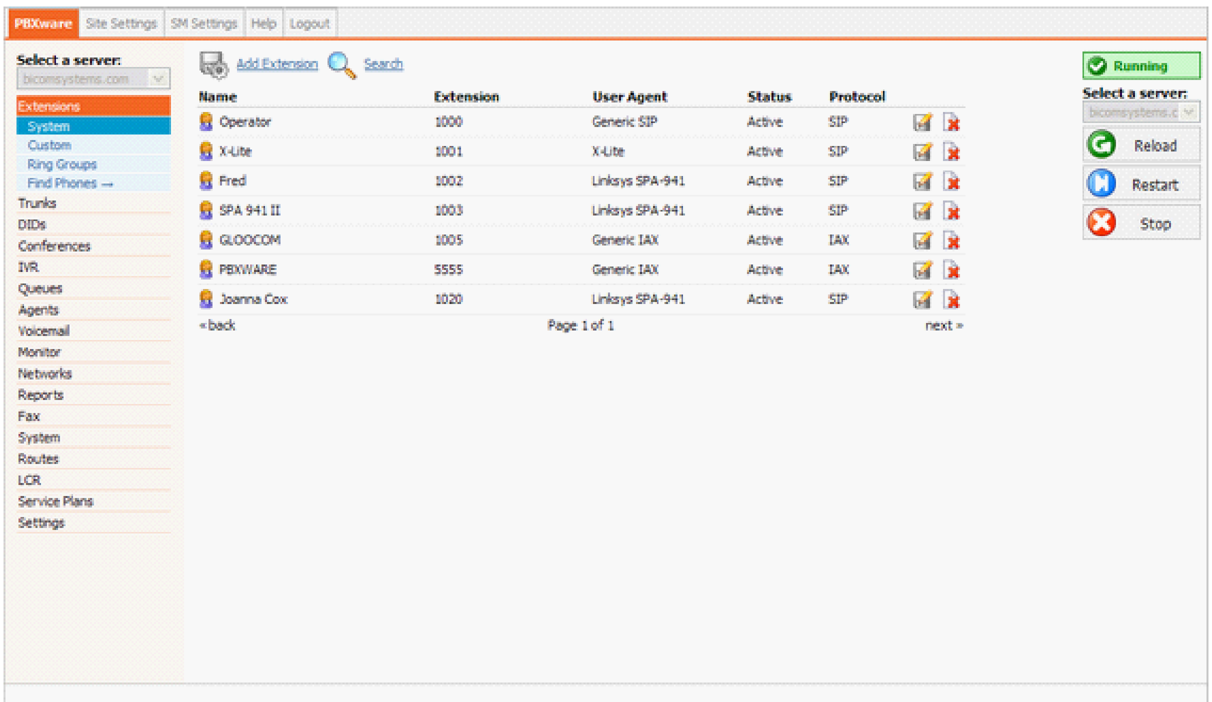

» Extensions: Edit
UAD: Generic SIP
Location: Local
Select User
Advanced Options
General
Extension: 1000
Name: Operator
E-mail: email@example.com
User Type: friend
DTMF Mode: rfc2833
Context: default
Status: active
Authentication
Username: 1000
Authname:
Secret: upfBapjl
PIN: 1662
Permissions
Destinations
Conferences
Enhanced Services
Call Monitoring
Save
Save & E-mail
Go back
Billing
Reset Inclusive
Credit/Debit
Billing: Yes No N/A
Service Plan: Euro
Slave: Yes No N/A
Slave Account Code:
Reminder Balance: 10.00000
Credit Limit: 100.00000
Package data (dd-mm-YYYY):
Expiry Date (dd-mm-YYYY):
Billing info
Account Balance: 100.00000
Available Funds: 200.00000
Inclusive Minutes Left: No Inclusives spent.
Creation Date: Unknown
First Use Date: Unknown
Last Use Date: Unknown
Network Related
NAT: Yes No Never Route N/A
Canreinvite: Yes No N/A
Qualify: 8000
Host: dynamic
Default IP:
Caller ID
Set Caller ID: Yes No N/A
Caller ID: Number: 1000 Name: Operator
CallerID Presentation: Please select..
Call Properties
Ringtime:
Incoming Dial Options:
Outgoing Dial Options:
VoiceMaster PIN:
Groups
Call Group:
Pickup Group:
Trunks
Primary Trunk: -- Default --
Secondary Trunk: -- Default --
Voicemail
Voicemail: Yes No N/A
Mailbox: 1000
Name: Operator
PIN: 1662
E-mail: email@example.com
Pager e-mail:
Greeting message: Unavailable
Reset Unavailable message: Yes No N/A
Reset Busy message: Yes No N/A
Skip Instructions: Yes No N/A
Attach: Yes No N/A
Delete After E-mailing: Yes No N/A
Say CallerID: Yes No N/A
Allow Review mode: Yes No N/A
Allow Operator: Yes No N/A
Operator Extension:
Play Envelope message: Yes No N/A
Voicemail Delay:
Timezone: Please select..
Speakerphone Page Auto-Answer SIP Header
Choose Device Type: Custom Header...
Custom Header:
Codecs
Disallow: all
Allow: G.711 μlaw G.711 alaw
G.723.1 G.726

Part IV

4 Extensions

Extensions are associated with all UADs/Phones connected to PBXware. Each UAD/Phone is assigned a network number, e.g. 1000. To reach other extensions, from any UAD/Phone registered with PBXware, just dial extension network number, e.g. 1022.

In this chapter we will cover following type of extensions:
- System 46
- Custom 97
- Ring Groups 100
- Find Phones 107



4.1 System

In this chapter we will cover:
- Search 51
- Add/Edit Extension 48
- Advanced Options 53

System extensions list all PBXware extensions with the following details:

Add Extension Search

Name	Extension	User Agent	Status	Protocol
John Smith	1000	Aastra 480i	Active	SIP
Peter Doyle	1006	Grandstream BT-102	Active	SIP
Joanna Cox	1008	Linksys SPA-941	Active	SIP
Joe Madsen	1016	Snom 320	Active	SIP
Opcom	1017	Generic IAX	Active	IAX
X-Lite	1003	X-Lite	Active	SIP
Jessica	1020	Linksys SPA-941	Active	SIP
1030	1030	Aastra 480i	Active	SIP
Terminal One	2001	Generic SIP	Active	SIP
Eyebeam	1009	Generic SIP	Active	SIP

« left Page 1 of 2 next »

Name:
Full name of the user device is registered to
Example: Peter Doyle
Field Type: Display

Extension:
UAD/Phone extension number
Example: 1111
Field Type: Display

User Agent:
UAD/Phone type
Example: Sipura SPA-841
Field Type: Display

Status:
UAD/Phone system status
Example: Active/Inactive
Field Type: Display

Protocol:
Protocol used by the UAD/Phone
Example: SIP/IAX
Field Type: Display

Edit UAD/Phone configuration
Example: Click to edit UAD/Phone configuration
Field Type: Button

Delete UAD/Phone from the system

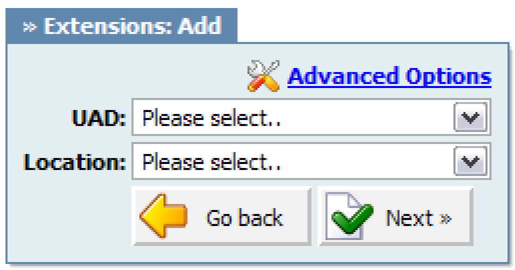
Example: Click to delete UAD/Phone from the system
Field Type: Button

4.1.1 Add/Edit Extension

Procedure for adding new system extension is divided into two steps. In first step, UAD/Phone type and extension location is provided. In second step, basic UAD/Phone information such as user's name and email address is provided.

NOTE: By default, 'Single Extension' will be created. 'Advanced Options' offer the facility to add multiple extensions as well. For more information please check the 'Adding Multi Extensions [52]' chapter.

Step One



UAD (User Agent Device)**:**

Select UAD/Phone model of the new extension. NOTE: In case that UAD/Phone is not listed here, please navigate to 'Settings: UAD', edit desired UAD/Phone and set 'Status' ='Active'.

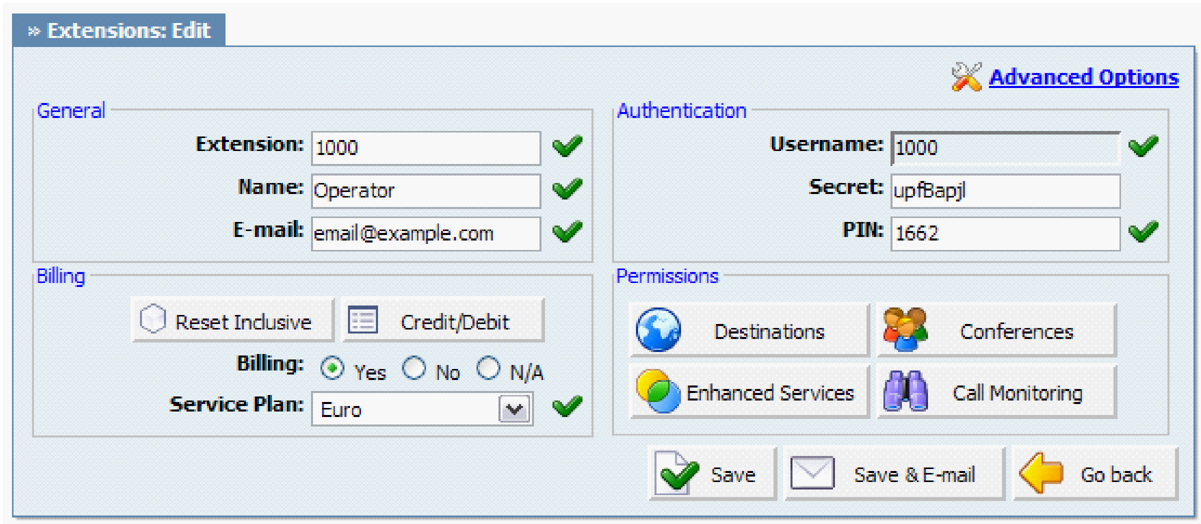
Example: If the new UAD/Phone you are adding to system is 'Linksys SPA-941', select it here
Field Type: Select box

Location:

Select the location of new UAD/Phone

Example: If the new UAD/Phone is located in local LAN, select 'Local', or 'Remote' if UAD/Phone is located outside of LAN
Field Type: Select box

Step Two



Extension:

System extension number

Example: Every system extension has its unique local network number. This number is dialed when trying to reach other system extension, e.g. 1000
Field Type: [0-9]

NOTE:
PBXware automatically fills this field by allocating the least unallocated number. To set different extension numbers allocating method navigate to 'Settings: Numbering Defaults' and apply new changes

By default extension number is 4 (four) digits long. The length of extension numbers is set at the initial state in setup wizard. To change the length of extension digits later, you must delete all PBXware end points that have extension number (e.g. extensions, ring groups, conferences, queues...) and set the new length at 'Settings: Servers: Edit: Number of digits' field. Please make sure to restart PBXware after made changes

Name:

Subscriber's full name

Example: Every PBXware extension is assigned to a subscriber. Provide subscriber name and last name here e.g. John Smith. This information is provided later, when making a call, as a caller id information and displayed on called party's UAD/Phone display
Field Type: [a-z][0-9]

E-mail:

Email address associated with the extension

Example: Email address provided here is used for various PBXware notifications (new voice mail received, unknown destination dialled, billing reminders, sending extension details...)
Field Type: [a-z][0-9]

Reset Inclusive:

Resets inclusive minutes

Example: An extension can be assigned to a certain billing service plan. Each service plan has its own set of prices for worldwide destinations, including inclusive minutes. Click this button in order to reset inclusive minutes and set them back to default for this extension.
Field Type: button

Credit/Debit:

Credit/debit extension

Example: Click on this button and manage extension credit/debit
Field Type: button

Billing:

Enable billing service

Example: In order to enable billing of all extension incoming and outgoing calls select 'Yes' here
Field Type: Option buttons

Service Plan:

Service plan used by extension

Example: PBXware can have multiple service plans, each with its own set of prices for worldwide destinations. Select here the service plan used by this extension
Field Type: Select box

Username:

Authentication username

Example: This option is used for UAD/Phone registration with PBXware. By default, this field is the same as the extension number and cannot be modified
Field Type: [0-9]

Secret:

Authentication password/secret

Example: This option is used for UAD/Phone registration with PBXware. By default this field is populated with random string, but this value can be changed to custom format at any time
Field Type: [a-z][0-9]

PIN (Personal Identification Number):

Extension authorization number

Example: This option is used to verify subscriber before critical UAD/Phone action is performed e.g. checking voicemail, performing remote PBXware access, accessing PIN protected enhanced services...). By default this field is populated with random integer, but this value can be changed to custom format at any time
Field Type: [0-9]

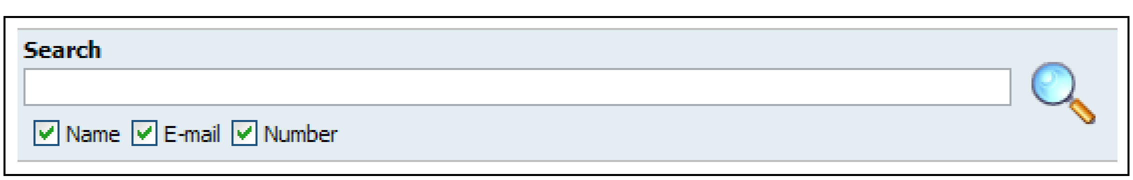
NOTE: PIN number must always be 4 (four) digits long

NOTE:

After extension is created, 'Permissions' group will be enabled and available for administration; 'Save & Email' button becomes available

4.1.1.1 Search

Search bar is very useful when dealing with many extensions. Extension details can be filtered by name, e-mail and extension number



Search:
Search phrase
Example: Provide a search phrase here and hit enter to filter the records
Field Type: [a-z][0-9]

Name:
Should search filter be applied to names UADs are registered to
Example: Check the box to search the names
Field Type: Check box

E-mail:
Should search filter be applied to email addressed associate with the UADs
Example: Check the box to search the email addresses
Field Type: Check box

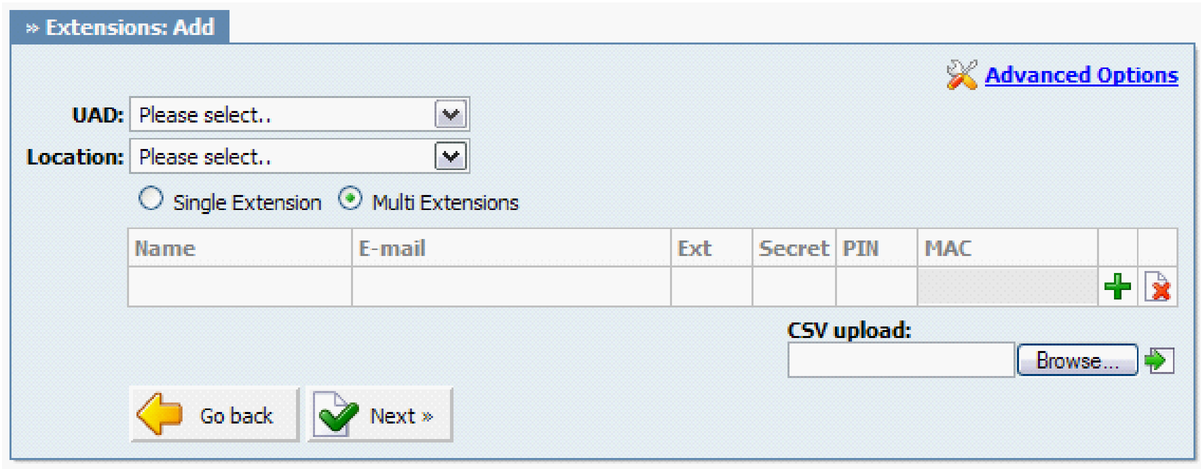
Number:
Should search filter be applied to extension numbers
Example: Check the box to search extension numbers
Field Type: Check box

4.1.1.2 Adding Multi Extensions

There are two ways to add multiple extensions to PBXware: by manually providing details for each extension and by uploading a .csv file with extension details

Manually
Provide details to 'Name', 'Email', 'Ext', 'Secret', 'PIN' [, 'MAC'] fields. Click on Add '+' button to add new extension or Delete 'x' button to delete extension details. When all extensions are entered click on the green Upload button to create extensions

Uploading '.csv' file
- Open text editor on your desktop
- Type extension details in following form ('Name', 'Email', 'Ext', 'Secret', 'PIN' [, 'MAC']). One extension per line.
  John Doe,john@domain.com,4444,1234,4444
  Joanna Cox,joanna@domain.com,5555,2345,5555
- Save file as 'ext.csv'
- Click on the 'Browse' button in PBXware interface
- Select 'ext.csv' from Desktop
- Click on 'Upload' button to create extensions

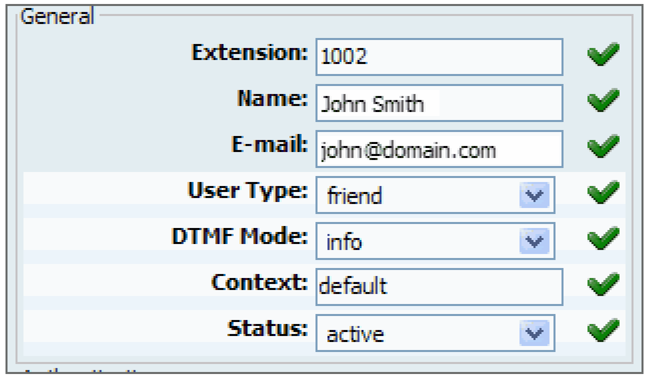


4.1.1.3 Advanced Options

In this chapter we will cover:

4.1.1.3.1 General Fields

Options for setting extension type, dtmf mode, context and status are most frequently used by UAD/Phone in their operation



User Type:

The way UAD/Phone is registered to PBXware

Example: Selecting 'User' will setup extension to make calls only, 'Peer' to receive calls only, and 'Friend' to make and receive calls
Field Type: Select box

DTMF Mode (Dual Tone Multi-Frequency) **:**

Set a tone frequency heard when pressing keys on phone keypad

Example: DTMF tones consists of two separate tones. Each key has a specific tone assign to it. When a key is pressed generated tone is sent to PBXware where it gets translated into a digit. In that way PBXware recognizes which number you are dialling. Not all countries/providers use the same DTMF mode. Select the appropriate tone here, as requested by service provider. If in doubt which option to use, select 'rfc2833' which is supported by most providers
Field Type: Select box

NOTE: You can test if DTMF is correctly set for UAD/Phone by checking extension voice mail '*123'. Provide a valid extension PIN when asked by PBXware. If you hear the 'Login incorrect' message, PIN entered was not recognized by PBXware. Select different DTMF mode here and try again until you login into extension voice box

Context:

A group to which extension belongs to

Example: Context is a collection/group of extensions. Each PBXware extension has to belong to a context. Unless set otherwise, each extension created is automatically assigned to 'default' context
Field Type: [a-z] [0-9]

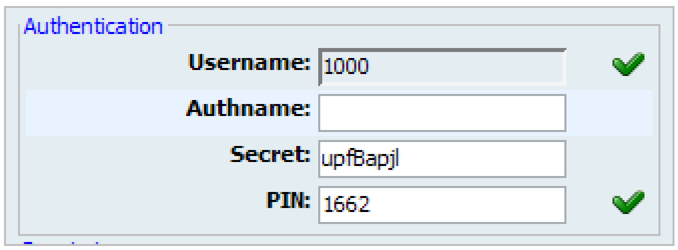
Status:

Extension's PBXware status

Example: Rather then deleting the extension and then recreating it again later on, extension can be activated/deactivated using this option. Setting this field to 'Not Active' will disable all calls to this extension.
Field Type: Select Box

4.1.1.3.2 Authentication

Just like a username and password is required when trying to access any remote computer, each UAD/Phone has to provide a valid username and password in order to register to PBXware.



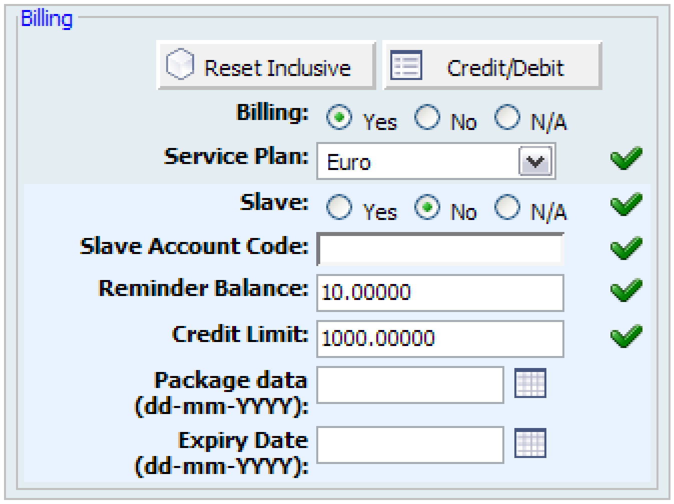
Authname:

Authentication name for some service providers

Example: Some service providers, due to security concerns, will require a special authentication name sent from extension (instead of extension number which is used by default). For example, if SIP provider requires '12lk3j4' username for authentication with their service, set '12lk3j4' here. The SIP request from UAD/Phone will be sent in following form '12lk3j4@sipprovider.com'
Field Type: [0-9][a-z]

4.1.1.3.3 Billing

PBXware has the ability to charge all incoming and outgoing extension calls. For billing system to work two things are needed: to create a service plan (with its set of prices) and to assign a service plan to extension. Following options are used to assign a service plan to extension and to manage extension billing funds



Slave:

Share extension billing funds with another extension

Example: Enabling this option for extension 1000 and setting 'Slave Account Code'='1001' allows these two extensions to share the same billing funds. Any call made by extension 1001 will take funds off extension 1000 billing plan. If no funds are available for extension 1000, neither of extensions will be able to make any calls
Field Type: Option buttons

Slave Account Code:

Extension which shares the billing funds

Example: This field is enabled when 'Slave' is set to 'Yes'. Provide extension number here, e.g. 1001, if you wish that extension 1001 shares the billing funds with edited extension
Field Type: [0-9]

Reminder Balance:

Account balance at which reminder is sent to extension owner

Example: If this field is set to 10, subscriber will receive email notification when extension billing funds reach this amount
Field Type: [0-9]

Credit Limit:

Maximum amount that PBXware can credit the subscriber

Example: This option is identical to bank account limit. Setting 'Credit Limit' to 10 will credit user the same

amount. Credit limit amount will be taken off the extension funds next time the funds are added.
Field Type: [0-9]

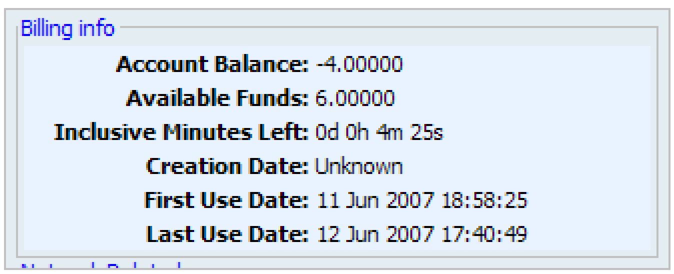
Package Date (dd-mm-YYYY):

Inclusive minutes expiry date

Example: If extension 1000 has used up all of its service plan inclusive minutes, and this field is set to 12-06-2007, inclusive minutes for this extension will be reset on this date, 12-06-2007. To reset inclusive minutes manually click on 'Reset Inclusive Minutes'
Field Type: [dd-mm-YYY]

4.1.1.3.4 Billing Info

Following options display extension billing status



Account Balance:

Sum spent by subscriber

Example: If subscriber has a 100 units in available funds, 100 will be displayed here. When negative value is displayed e.g. -49, it means that extension funds are 0 and 49 units of credit limit has been used
Field Type: Display

Available Funds:

Available funds (account balance + credit limit)

Example: If subscriber has a 100 units in available funds and 10 units in credit funds, 110 is displayed here
Field Type: Display

Inclusive Minutes Left:

Inclusive Minutes Left

Example: As long as there is any inclusive time left, billing is not calculated for outgoing calls. Inclusive time left is displayed here e.g. 0d 0h 4m 25s
Field Type: Display

Creation date:

Date PBXware extension was created

Example: 14-06-2007 12:30:36
Field Type: Display

NOTE: If PBXware was updated to 2.0 or newer version, old extensions will have this field displaying 'unknown' and all new extensions will display extension creation date

First Use Date:

Date/Time of the first extension use

Example: 11 Jun 2007 18:58:25
Field Type: Display

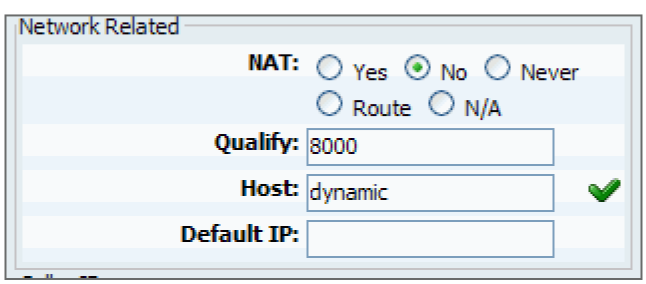
Last Use Date:

Date/Time of the last extension use

Example: 11 Jun 2007 19:25:12
Field Type: Display

4.1.1.3.5 Network Related

These options set important network related features such as the NAT status and IP address from which UAD/Phone registers to PBXware



NAT (Network Address Translation):

Tell PBXware if extension is located behind NAT

Example: If extension 1000 is behind NAT in a remote location/network set this option to 'Yes'. If this option is not set to 'Yes' you may experience a 'one way audio problem'. Options to choose from:

- **Yes** - Remote extension is behind NAT
  - **No** - Remote extension is not behind NAT (default value)
  - **Never** - Never attempt NAT mode
  - **Route** - Assume NAT but do not send rport

Field Type: Option buttons

Qualify:

Timing interval at which PBXware checks UAD/Phone availability

Example: This timing interval is set in milliseconds and it behaves like a 'ping' command. A request is send by PBXware and reply is expected from UAD/Phone. Default value for this option is 2500 (2.5 seconds)
Field Type: [0-9]

NOTE: To see availability of all PBXware extensions navigate to 'Monitor: Extensions' and check the 'Status' field which should be set to 'online/offline'. If 'unknown' is displayed, it means that this 'Qualify' value has not been set for the extension

Host:

UAD/Phone IP address

Example: If UAD/Phone is on a static IP set its IP address here e.g. 192.168.1.22. In case that UAD/Phone is on dynamic IP address, set 'dynamic' here
Field Type: [dynamic][0-9]

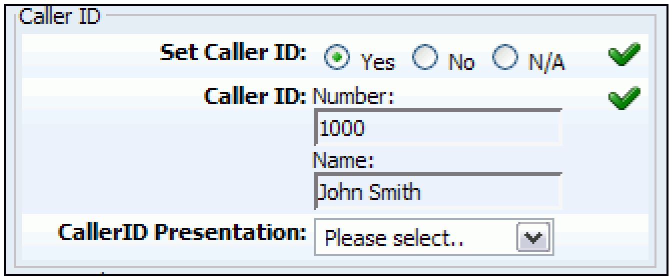
Default IP:

IP address used until UAD/Phone registers with PBXware

Example: This IP address, e.g. 213.22.30.200, will be used by PBXware until UAD/Phone registers on 'Host' IP address
Field Type: [0-9]

4.1.1.3.6 Caller ID

These options identify caller by displaying his name and extension number on called party's UAD/Phone display. Note that following options are read only (taken from 'Name' and 'Number' fields). In order to change caller id information navigate to 'Enhanced Services: Caller ID'



Set Caller ID:

Enable caller id service

Example: Selecting 'Yes' will enable the caller id service
Field Type: Option buttons

Caller ID:

Caller id name and number information

Example: Note that following options are read only (taken from 'Name' and 'Number' fields). In order to change the caller id information navigate to 'Enhanced Services: Caller ID'
Field Type: Read-only

Caller ID Presentation:

Change the presentation for the caller id information

Example: If experiencing problems with passing the caller id information to called party, select one of the presentation options here and place a call again. Issues with caller id information not being passed usually happen when placing calls over trunks. Before making any changes here, please make sure that service provider supports custom caller id information
Field Type: Select box

4.1.1.3.7 Voicemail

These options mimic the functions of an answering machine but with many additional features added. Each extension created by default has an active mailbox with the same extension number. Note that all PBXware voice messages are saved on central file-system location instead of UAD/Phone

Accessing voice-box:

To access a voice-box dial '*123', enter extension PIN number and follow the instructions

Leaving a voice message:

When transferred to a voice-box user will hear '...Please leave your message after the tone. When done, hangup or press the # key' message.

User has two options available:
1. Leave a voice message(ended by pressing '#' key or by hanging up), or
2. Reach an operator by dialing '0'

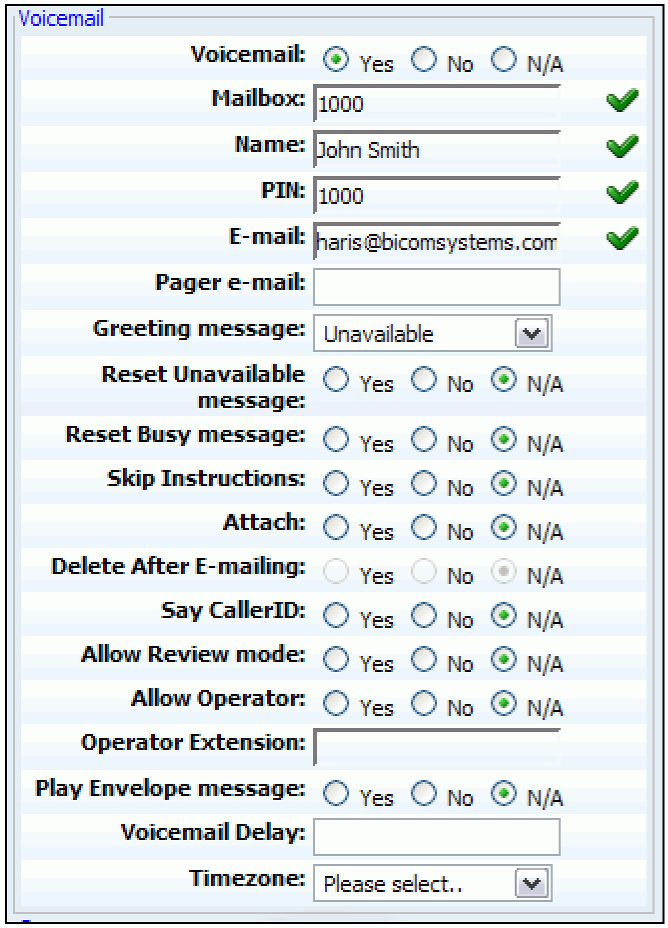
If '0' is dialed, 'Press 1 to accept this recording, otherwise please continue to hold' message is heard

User has two options:
1. Press '1' to save a message, after which the operator will be dialed. 'Please hold while i try that extension' will be heard, or
2. Continue to hold, which will delete a voice message, after which the operator will be dialed. 'Message deleted, please hold while i try that extension' will be heard.

Voice mail filesystem usage:

With continuous tone 60 seconds:
- wav49 = 91.0kb
- wav = 863.0kb
- gsm = 91.0kb

With continuous silent tone 60 sec:
- wav49 = 0.38kb
- wav = 3.0kb
- gsm. = 0.32k b



Voicemail:

Enable the voicemail service

Example: Enabling this option will create extension voicemail box
Field Type: Option buttons

Mailbox:

Mailbox number

Example: Every system mailbox has its unique local network number. This option is the same as the extension number , e.g. 1000
Field Type: Readonly

Name:

Subscriber's full name

Example: Every PBXware voicemail box is assigned to a subscriber. This option displays subscriber name, is the same as the 'General: Name' field
Field Type: Readonly

PIN (Personal Identification Number):

Voicemail authorization number

Example: This option is used to verify subscriber before entering voice box and is the same as the 'General: PIN' field
Field Type: Readonly

E-mail:

Email address associated with the voice box

Example: Email address provided here is used for voicemail notifications and is the same as the 'General: E-mail' field
Field Type: [a-z] [0-9]

Pager e-mail:

Pager e-mail address associated with the voice box

Example: Pager email address provided here is used for new voicemail notification
Field Type: [a-z] [0-9]

Greeting message:

Type of greeting message played to callers transferred to voice box

Example: Select between 'Unavailable' and 'Busy' greeting message here.
Field Type: Select box

NOTE:
Default unavailable message = 'The person at extension $NUMBER is unavailable. Please leave your message after the tone. When done hang up or press the # key'

Default busy message = 'The person at ext $NUMBER is on the phone. Please leave your message after the tone. When done hang up or press the # key'

To record custom unavailable or busy message, enter voice box (*123), press 0 for mailbox options and then press 1 to record unavailable or 2 to record BUSY message

Reset Unavailable message:

Resets custom recorded unavailable message

Example: If extension uses custom recorded unavailable message, select 'Yes' here and click on 'Save' button to reset it back to default one
Field Type: Option buttons

Reset Busy message:

Resets custom recorded busy message

Example: If extension uses custom recorded busy message, select 'Yes' here and click on 'Save' button to reset it back to default one
Field Type: Option buttons

Skip Instructions:

Do not play instructions on how to leave a voice message

Example: By default, when user is transferred to a voice box, instructions on how to leave a voice message are played 'Please leave your message after the tone. When done hang up or press the # key'. Select 'Yes' disable/skip these instructions.
Field Type: Option buttons

Attach:

Attach the voice message as a sound file to a voicemail notification email

Example: If this option is enabled, a sound file with a recorded message will be attached to a voicemail notification email
Field Type: Option buttons

Delete After E-mailing:

Should voice mail be deleted from the file system after sending it as an attachment in notification email

Example: If this option is enabled, a sound file with a recorded voice message will be deleted from the file system after being sent as an attachment in notification email
Field Type: Option buttons

Say Caller ID:

Should PBXware announce extension number from which the voice message has been recorded when checking voice box

Example: If this option is enabled, when checking voice box, 'From phone number {$NUMBER}' message will be heard
Field Type: Option buttons

Allow Review mode:

Should callers be permitted to review their voice messages before committing them to voice box

Example: If this option is disabled and user after leaving a voice message presses the '#' key, voice message is committed to voice box. But if this option is enabled, after pressing the '#' key user is offered three options:

- 1 to accept this recording
- 2 to listen to it
- 3 to re-record your message

Select preferred option and follow the instructions
Field Type: Option buttons

Allow Operator:

Should callers be allowed to reach an operator from the voice box options

Example: If this option is enabled, and user after leaving a voice message presses the '#' key, 'Press 0 to reach an operator' will be heard. Once user dials 0, extension set as 'Operator Extension' will ring
Field Type: Option buttons

Operator Extension:

Local extension number that acts as an operator

Example: If 'Allow Operator' option is enabled, and user after leaving a voice message presses the '#' and '0' keys, extension set here, to act as an operator, will ring
Field Type: [0-9]

Play Envelope message:

Should date/time and extension number from which the message was recorded be announced

Example: If this option is enabled 'Received at $DATE, from phone number $NUMBER' message will be heard when checking voice messages
Field Type: Option buttons

Voicemail Delay:

Pause in seconds made by PBXware before asking user for PIN/Password

Example: Some UADs/Phones have tendency to garble the beginning of a sound file. Therefore, user

checking the voice box when asked for password might hear '...sword' instead of 'Password'. Setting this field to '1' or '2' will provide one or two seconds of silence before playing the 'Password' message without missing the beginning of file
Field Type: [0-9]

Timezone:

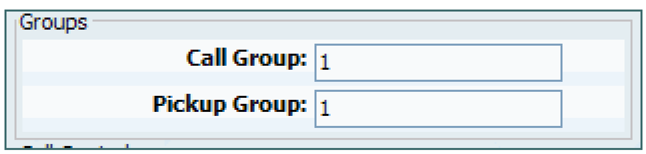
Sets the correct date/time stamp

Example: By setting the correct time zone, user would always be notified of the exact date/time voice message was left on their box. Set the correct time zone if user is located in different time zone then PBXware.
Field Type: Select box

NOTE: Time zones are taken from '/usr/share/zoneinfo' system directory

4.1.1.3.8 Groups

These options define who is allowed to pickup our incoming calls, and whose calls we are allowed to pickup



Call Group:

The call group this extension belongs to

Example: In a way similar to context grouping, only this option sets to which call group extension belongs to. Extension can be assigned to more call groups. Use commas (1,2,3...) or dashes(1,2-7) to assign an extension to two or more call groups. Note that allowed range is 0-63.
Field Type: [0-9] [,-]

NOTE: Grouping works only within a technology(SIP to SIP or IAX to IAX).

Pickup Group:

Call groups this extension is allowed to pickup

Example: In a way similar to context grouping, only this option sets which call groups this extension is allowed to pickup. Extension can be assigned to more call groups. Use commas (1,2,3...) or dashes(1,2-7) to assign an extension to two or more call groups. Note that allowed range is 0-63.
Field Type: [0-9] [,-]

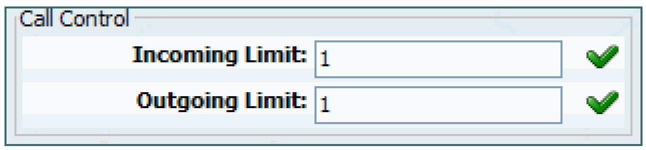
NOTE:

Grouping works only within a technology(SIP to SIP or IAX to IAX).

If extension 1000 belongs to call group 2, and extension 1005 has pickup group set to 2; to pickup a call from ringing extension 1000, dial *8 from extension 1005. If more call group extensions are ringing at the same time dial '*88 + $EXTENSION' to pickup specific ringing extension

4.1.1.3.9 Call Control

These options set the number of simultaneous incoming/outgoing extension calls



Incoming Limit:

Maximum number of simultaneous incoming calls

Example: If extension receives more incoming calls then set here, they are all redirected to extension voice-box
Field Type: [0-9]

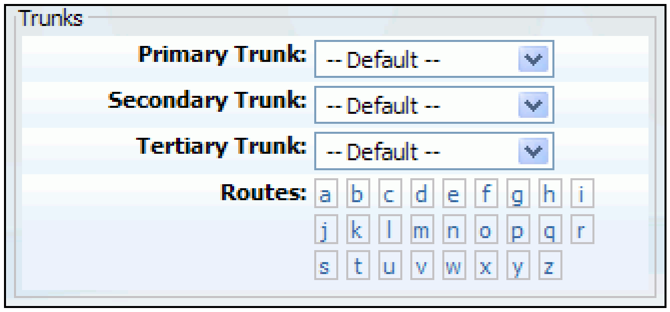
Outgoing Limit:

Maximum number of simultaneous outgoing calls

Example: With some UADs/Phones (Linksys SPA-941 for example) a call can be put on hold on line one, another call can be made on line two and PBXware will treat them as one call. On other UADs/Phones (EyeBeam for example) when one call is on hold, another one cannot be made when this, 'Outgoing Option' is set to 1
Field Type: [0-9]

4.1.1.3.10 Trunks

These options allow extension to use its own default trunks and routes for all outgoing calls



Primary/Secondary/Tertiary Trunk:

Default trunks for all routes dialed from extension

Example: If using these options, it is recommended that all three trunks are selected. If primary trunk cannot deliver the call, secondary trunk is used etc...
Field Type: Select box

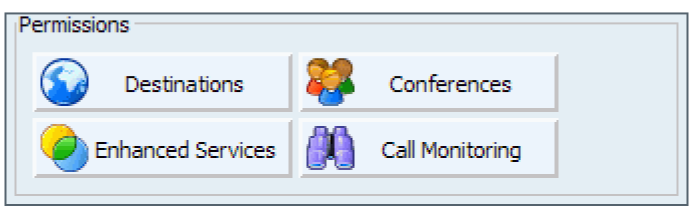
NOTE: These options override default trunks on system level 'Settings: Default Trunks'

Routes:

Click here for more

4.1.1.3.11 Permissions

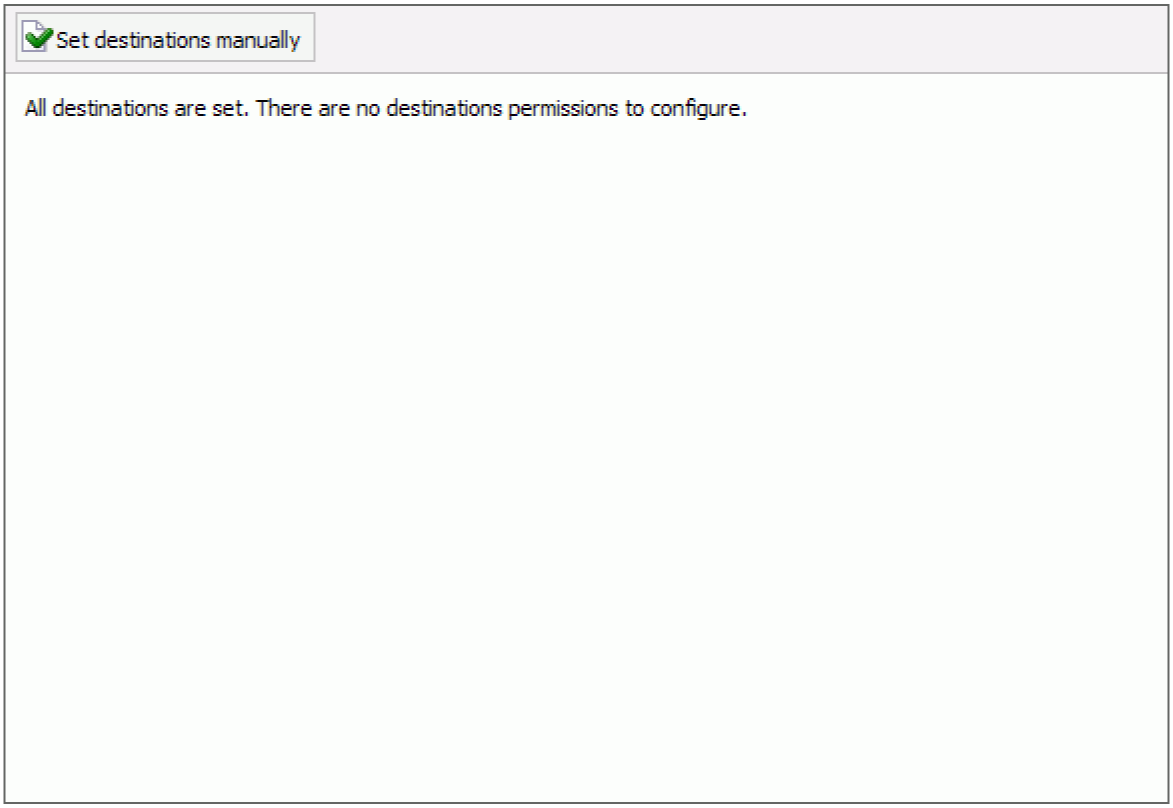
In this chapter we will cover:



4.1.1.3.11.1 Destinations

These options grant/deny access to remote, local and other network destinations. Remote destinations are all destinations outside of PBXware network (proper/mobile numbers). Local destinations are all available PBXware extensions (extensions, conferences, ivrs, queues...). Other network destinations are other PBXware networks.

When 'Destinations' button is clicked, and window bellow is displayed, it means that all remote, local and other network destinations are allowed for this extension. In order to edit the destinations manually click on 'Set destinations manually' button.

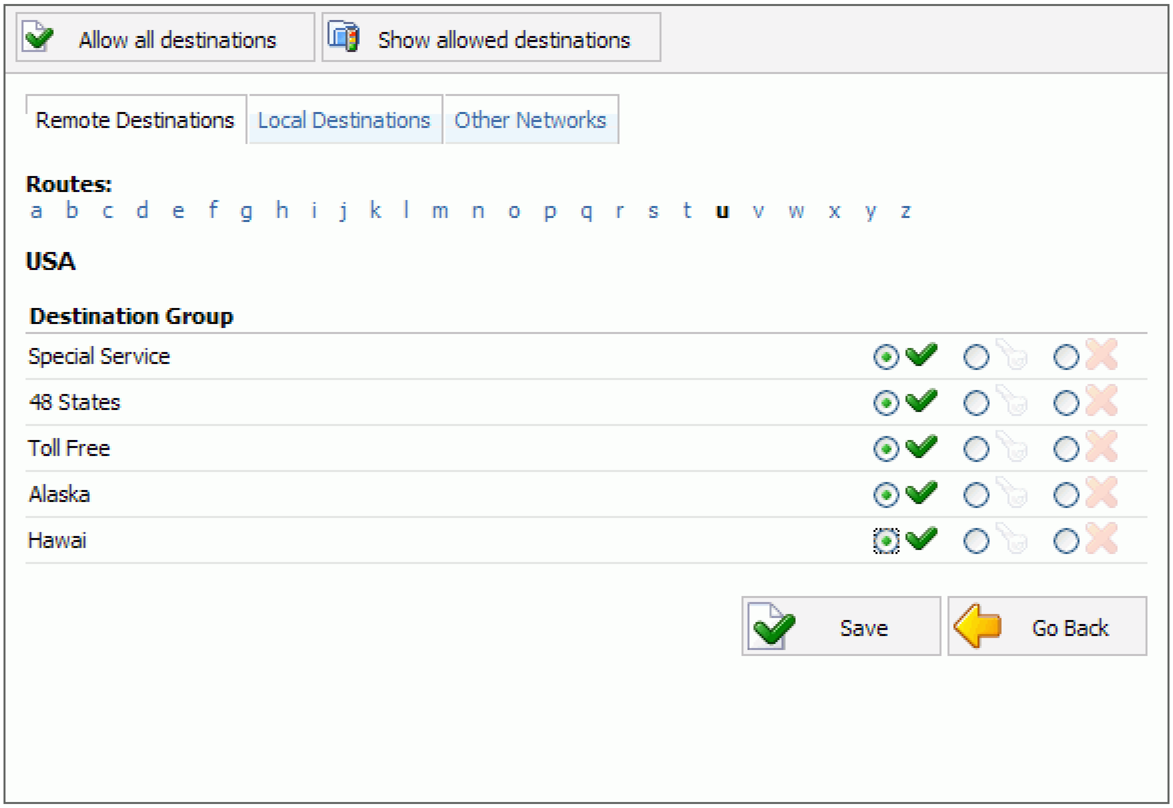


When 'Set destinations manually' button is selected all available remote, local and other network destinations are displayed. Set preferred destination permissions and click on 'Save' button when done. You may choose between following permissions:

 Authorized

 PIN Required

 Not Authorized



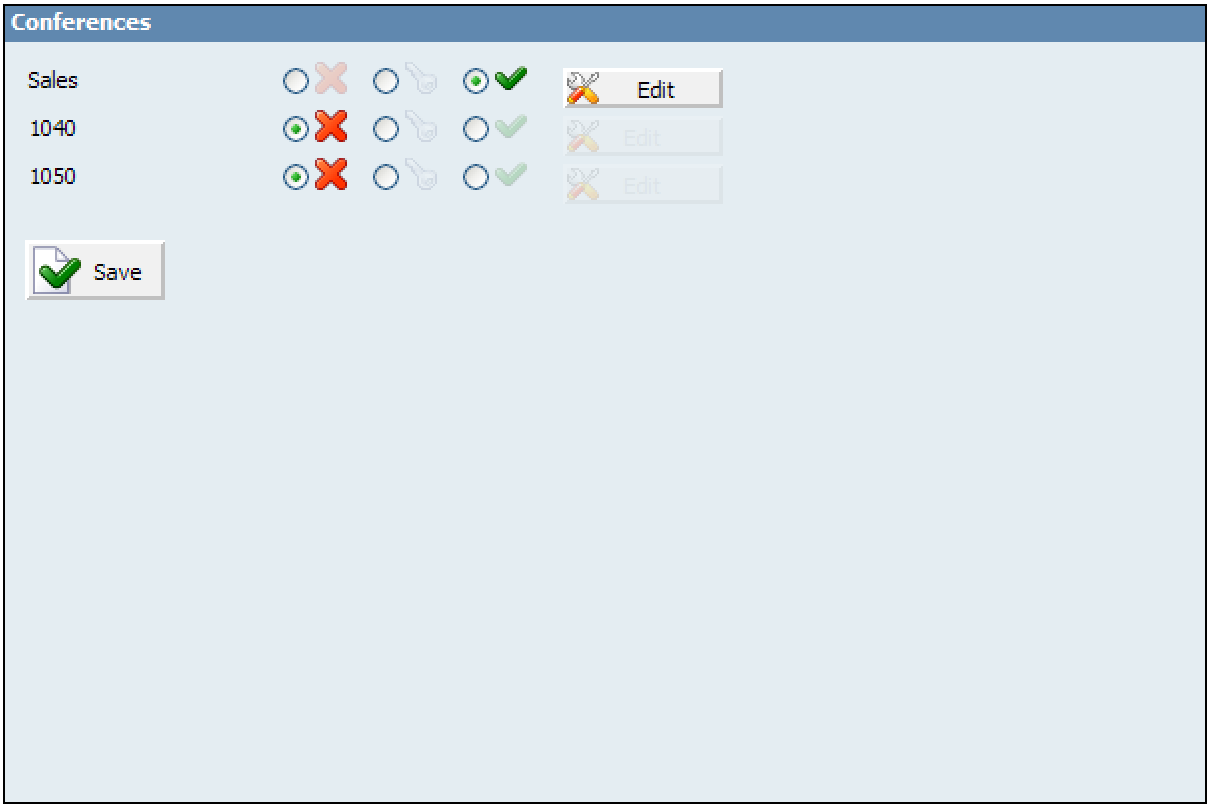
4.1.1.3.11.2 Conferences

These options grant/deny access to PBXware conferences. You may choose between following permissions:

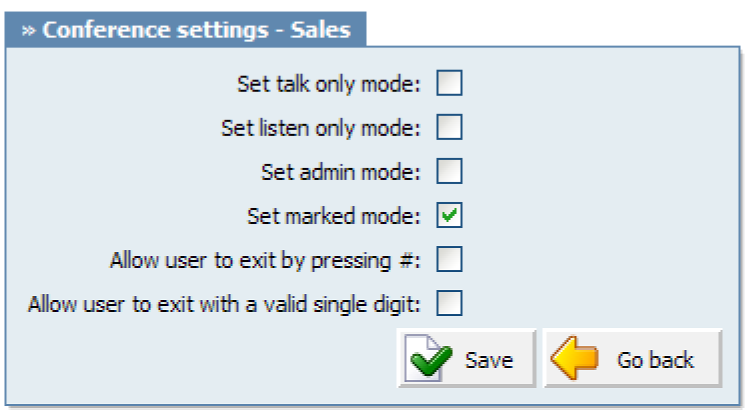

Not Authorized


PIN Required


Authorized



These options apply specific conference rules to extension



Set talk only mode:

Sets the talk only conference mode

Example: With this option enabled, conference calls coming from this extension will be allowed to talk only (no voice will be heard on the UAD/Phone handset)
Field Type: Check box

Set listen only mode:

Sets the listen only conference mode

Example: With this option enabled, conference calls coming from the extension will be allowed to listen only (no voice will be sent from the UAD/Phone handset)
Field Type: Check box

Set admin mode:

Sets the admin conference mode

Example: With this option enabled, conference calls coming from the extension will be treated with admin privileges
Field Type: Check box

Set marked mode:

Sets the marked conference mode

Example: With this option enabled, conference calls coming from the extension will be treated with less then admin, but higher then regular conference participants privileges
Field Type: Check box

Allow user to exit by pressing #:

Example: With this option enabled, user will be allowed to exit the conference by dialing the '#' key
Field Type: Check box

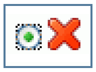
Allow user to exit with a valid single digit:

Example: With this option enabled, user will be allowed to exit the conference by dialing any digit
Field Type: Check box

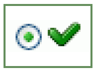
4.1.1.3.11.3 Enhanced Services

These options grand/deny access to enhanced extension features such as custom caller id information, call pickup etc... Enhanced services are ordered by priority (marked by numbers 01, 02...). Higher the priority number equals higher precedence of the enhanced service. For example, due to similarity of 'Follow Me' and 'Group Hunt' services, if both enabled, 'Follow Me' will be executed due to higher precedence

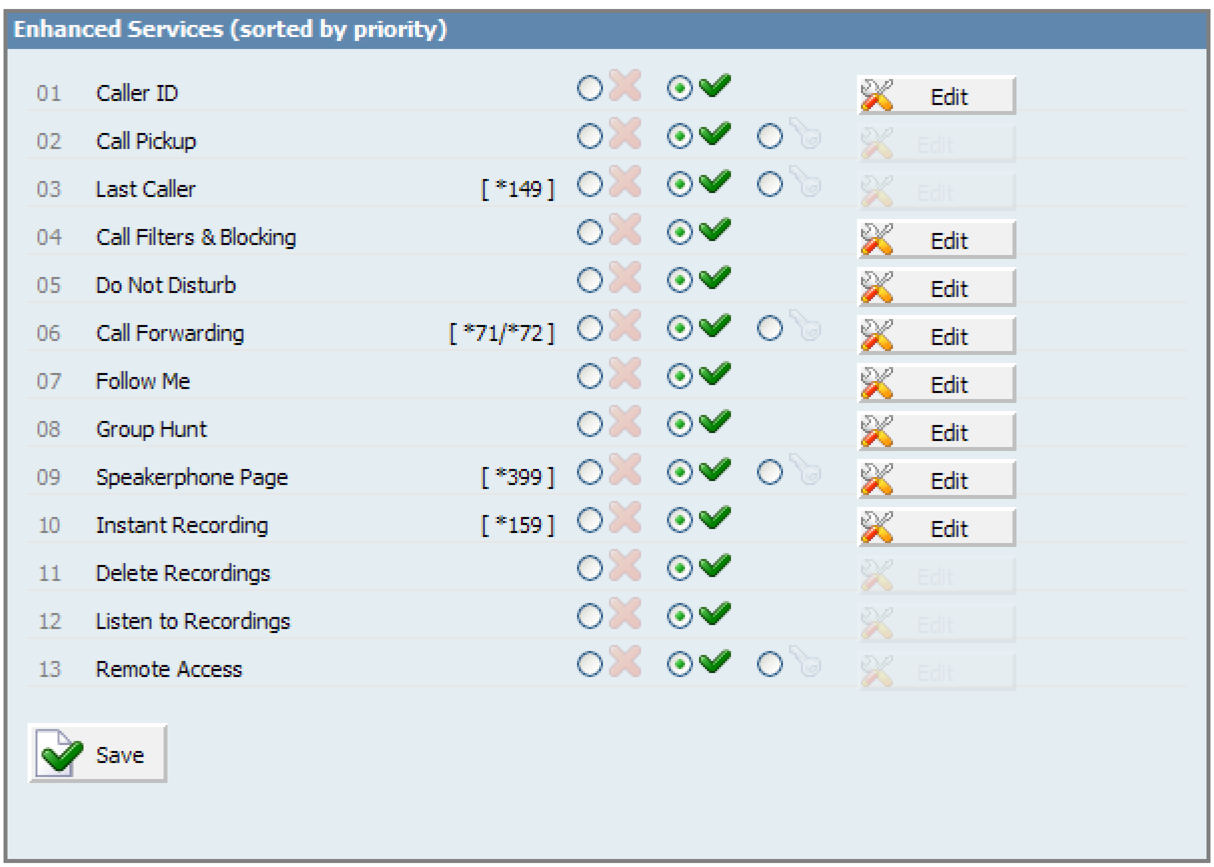
You may choose between following enhanced services permissions:


Not Authorized


PIN Required


Authorized

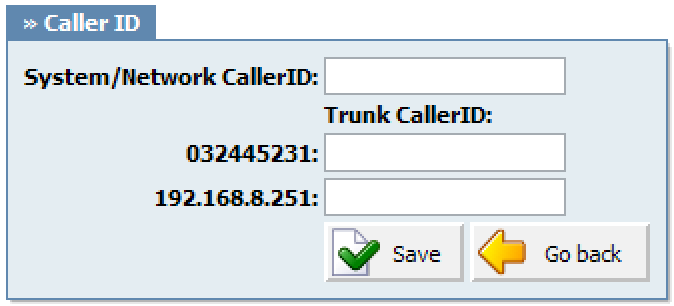
In this chapter we will cover:


Enhanced Services (sorted by priority)

01	Caller ID		Edit
02	Call Pickup		Edit
03	Last Caller	[*149]	Edit
04	Call Filters & Blocking		Edit
05	Do Not Disturb		Edit
06	Call Forwarding	[*71/*72]	Edit
07	Follow Me		Edit
08	Group Hunt		Edit
09	Speakerphone Page	[*399]	Edit
10	Instant Recording	[*159]	Edit
11	Delete Recordings		Edit
12	Listen to Recordings		Edit
13	Remote Access		Edit

Save

These options set custom caller id information (displayed on called party UAD/Phone display). Caller id can be set differently for local extensions and each of the trunks



System/Network Caller ID:

Caller id information displayed on local and other PBXwares in a network

Example: Setting this option to 'JaKe 2oo7' and calling system extension 1005, or extension 1007 on other PBXware in a network, will display 'JaKe 2oo7' on both UAD/Phone displays
Field Type: [a-z][0-9]

Trunk CallerID:

Caller id information displayed per trunk

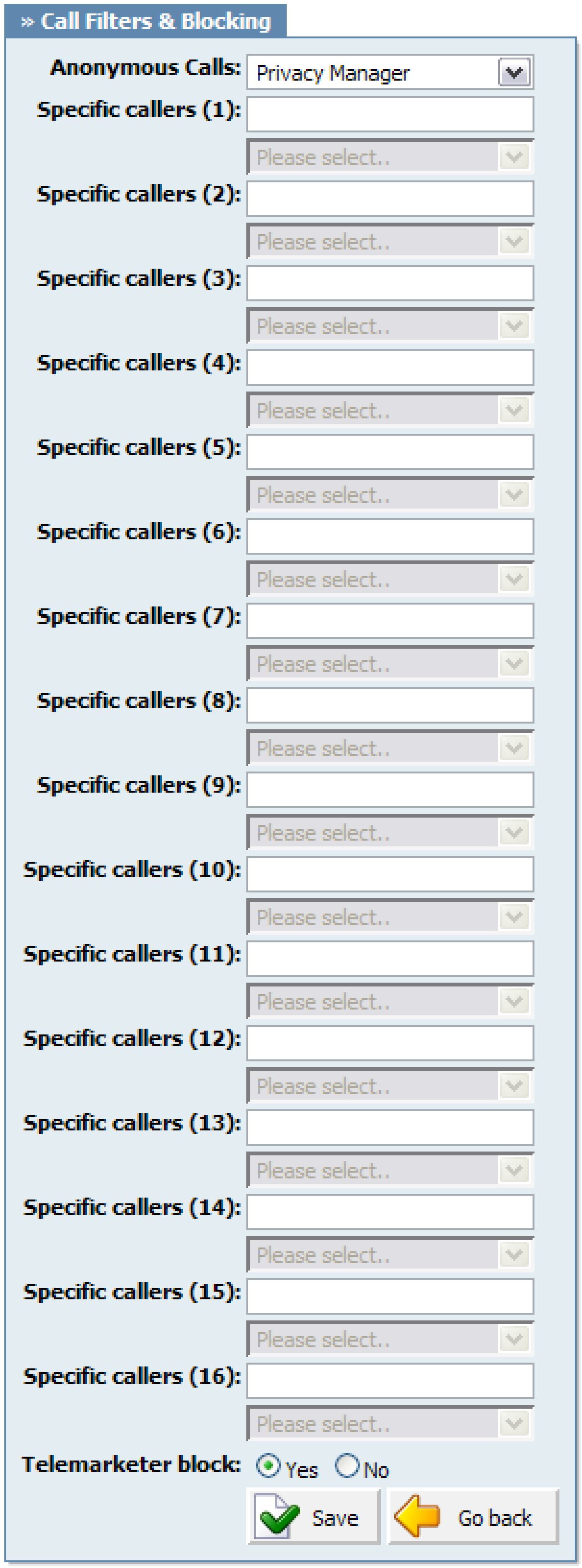
Example: Setting this option to 'JaKe 2oo7' for trunk '032445231' will display 'JaKe 2oo7' on any UAD/Phone reached over this trunk. NOTE: Custom caller id information must be allowed by the service provider
Field Type: [a-z][0-9]

This option enables user to pickup ringing calls of the same call group (See 'Extensions: Groups 66' for more)

This option enables user to dial back the last extension that was calling him. For example, dial '*149' to hear the number of the last extension calling you and press '0' to call back that number

NOTE: Access code for this service can be customized through 'Settings: Access Codes'

These options filter and block all incoming calls based on a custom set of rules. For example, all anonymous calls can be blocked or redirected to voice box...


» Call Filters & Blocking
Anonymous Calls: Privacy Manager
Specific callers (1):
Please select..
Specific callers (2):
Please select..
Specific callers (3):
Please select..
Specific callers (4):
Please select..
Specific callers (5):
Please select..
Specific callers (6):
Please select..
Specific callers (7):
Please select..
Specific callers (8):
Please select..
Specific callers (9):
Please select..
Specific callers (10):
Please select..
Specific callers (11):
Please select..
Specific callers (12):
Please select..
Specific callers (13):
Please select..
Specific callers (14):
Please select..
Specific callers (15):
Please select..
Specific callers (16):
Please select..
Telemarketer block: Yes No
Save
Go back

Anonymous Calls:

Action taken on anonymous calls (calls without caller id information)

Example: If set to 'Always Busy', all calls marked as 'Anonymous' (without caller id information) will hear the busy signal
Field Type: Select option

Specific callers:

Action taken when specific number dials

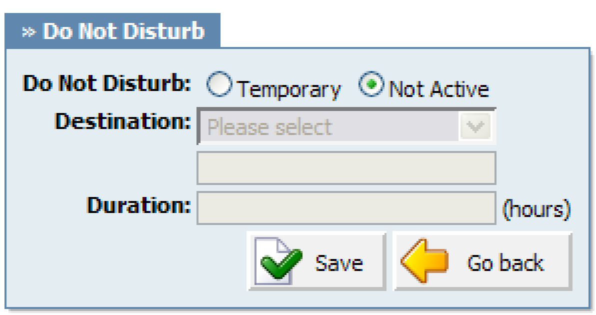
Example: If set to '1000' and action taken is 'Busy', all extensions will be able to call in except extension 1000, which will hear the busy sound
Field Type: Select option

Telemarketer block:

Block all telemarketer calls

Example: Set to 'Yes' to block the telemarketer calls
Field Type: Select option

These options temporarily redirect all incoming calls to a preset destination (extension number or voice box). This service is time based and is set in hours



Do Not Disturb:

Activate the DND service

Example: This service can be set only on a temporary basis. Select 'Temporary' to activate it

Field Type: Option buttons

Destination:

Destination to be dialed once this service is enabled

Example: Select 'Call Forward' here and provide (local/mobile/proper) number in the field bellow to redirect all calls to; or select 'Voicemail' and set the voicemail box number
Field Type: [0-9]

Duration:

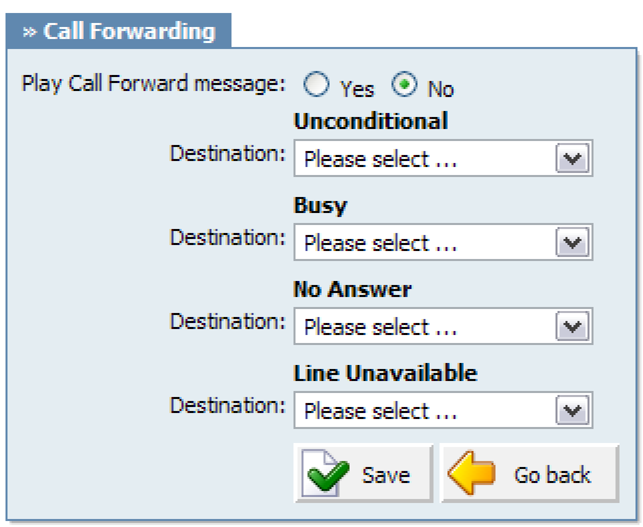
Time in hours the service will be active

Example: Set '1' to enable the service for one hour
Field Type: [0-9]

These options forward all incoming calls based on custom set of rules. For example, calls can be forwarded to other extensions (local/proper/remote) or voicemail boxes unconditionally or depending on whether extension is busy, nobody answers or when line is unavailable.

To unconditionally forward all calls to extension 1000, from your extension dial '*71+1000'. To disable unconditional call forwarding dial '*72'.

NOTE:
Some UADs/Phones (Linksys SPA-941) use *72 for additional UAD/Phone features.
Access code for 'Unconditional Call Forwarding' can be customized through 'Settings: Access Codes'.



Play Call Forwarding Message:

Notify users their call is being forwarded

Example: If call is forwarded to proper/mobile numbers for example, callers should be notified about it. With this option enabled, caller would hear 'Please wait, your call is being forwarded. You're not being charged for the forwarding part of the call' message
Field Type: Option buttons

Unconditional:

Forward all incoming calls

Example: With this option enabled, all incoming calls are redirected to number/voice box set in the field bellow e.g. 1000
Field Type: Option buttons

Busy:

Forward incoming calls if extension is busy

Example: With this option enabled, all callers trying to reach this extension when the same is busy will be forwarded to number/voice box set in the field bellow e.g. 1000
Field Type: Option buttons

No Answer:

Forward incoming calls if nobody answers

Example: With this option enabled, if nobody answers the incoming call the same is forwarded to number/ voice box set in the field bellow e.g. 1000
Field Type: Option buttons

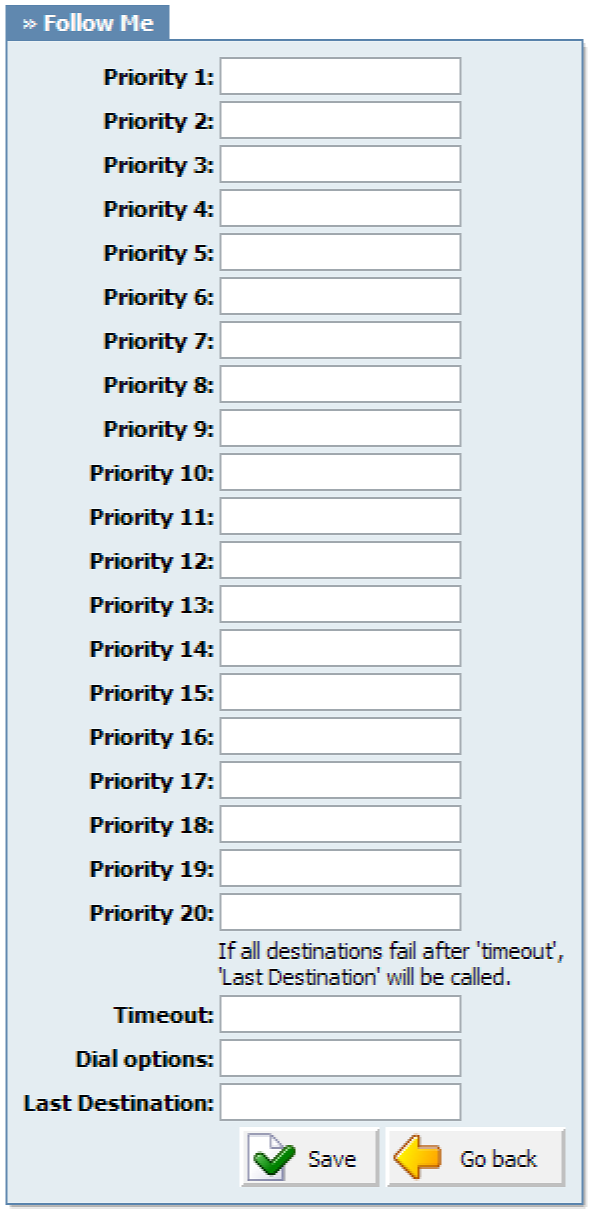
Line Unavailable:

Forward incoming calls if line is unavailable

Example: With this option enabled and if this extension is unavailable (not registered with the PBXware) all calls will be forwarded to number/voice box set in the field bellow e.g. 1000
Field Type: Option buttons

These options ring multiple extensions in sequential order

NOTE: If placing calls to mobile/proper number it may take 2-3 seconds until call is placed over Zaptel



Priority *:

Local/Proper/Mobile numbers to be dialed

Example: Provide a list of local/proper/mobile extensions that are to be dialled in sequential order when this extension is called. If first extensions does not answer the call, second one rings. If none of the 'Priority' extensions answers the call, caller gets redirected to 'Last Destination' extension
Field Type: [0-9]

Timeout:

Ring time in seconds

Example: Setting '20' here will ring each of the 'Priority' extensions for 20 seconds. If first extension does not answer during this timeout period, second extension is called etc...
Field Type: [0-9]

Last Destination:

The last destination to be dialled

Example: Extension number dialled if none of the 'Priority' extensions answers the call. In case that this 'Last Destination' does not answer the call, the call is redirected to 'Last Destination' voice box
Field Type: [0-9]

Dial Options:

Additional call properties

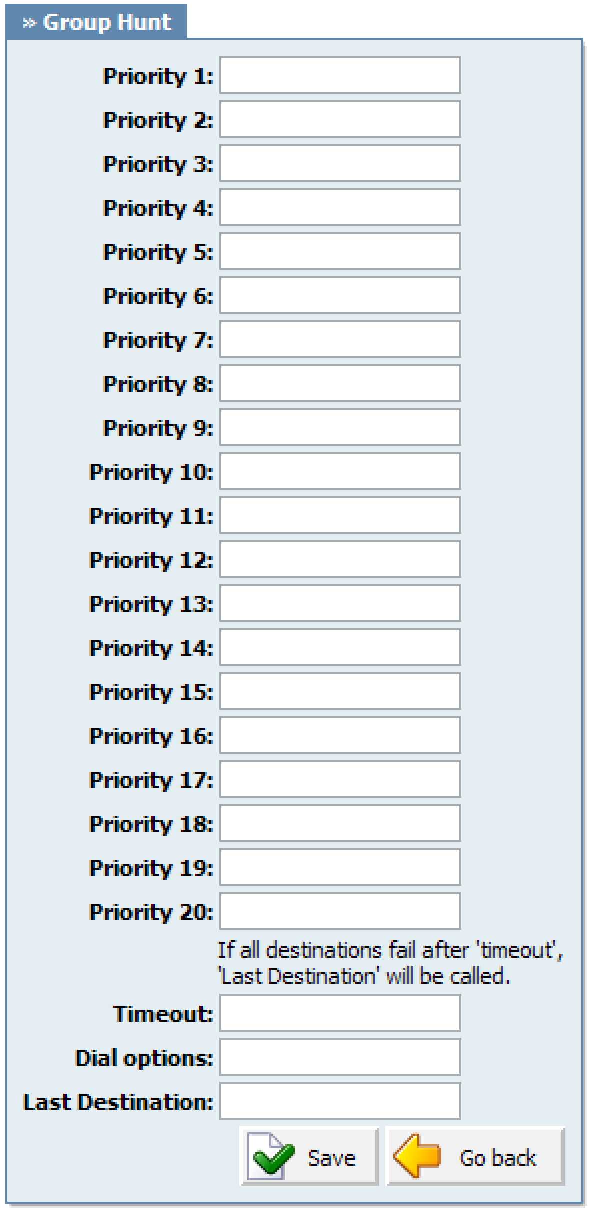
Example: These options additionally customize the call properties such as allowing the called party to transfer the call, to play MOH music instead of the ringtone etc...
Field Type: [a-z]

- **t** - Allow the called user to transfer the call by hitting #
- **T** - Allow the calling user to transfer the call by hitting #
- **r** - Generate a ringing tone for the calling party, passing no audio from the called channel(s) until one answers. Use with care and don't insert this by default into all your dial statements as you are killing call progress information for the user. Really, you almost certainly do not want to use this. Asterisk will generate ring tones automatically where it is appropriate to do so. 'r' makes it go the next step and additionally generate ring tones where it is probably not appropriate to do so.
- **R** - Indicate ringing to the calling party when the called party indicates ringing, pass no audio until answered. This is available only if you are using kapejod's bristuff.
- **m** - Provide Music on Hold to the calling party until the called channel answers. This is mutually exclusive with option 'r', obviously. Use m(class) to specify a class for the music on hold.
- **o** - Restore the Asterisk v1.0 Caller ID behaviour (send the original caller's ID) in Asterisk v1.2 (default: send this extension's number)
- **j** - Asterisk 1.2 and later: Jump to priority n+101 if all of the requested channels were busy (just like behaviour in Asterisk 1.0.x)
- **M(x)** - Executes the macro (x) upon connect of the call (i.e. when the called party answers)
- **h** - Allow the callee to hang up by dialing *
- **H** - Allow the caller to hang up by dialing *
- **C** - Reset the CDR (Call Detail Record) for this call. This is like using the NoCDR command
- **P(x)** - Use the Privacy Manager, using x as the database (x is optional)

- **g** **-** When the called party hangs up, exit to execute more commands in the current context.
- **G(context^exten^pri) -** If the call is answered, transfer both parties to the specified priority; however it seems the calling party is transferred to priority x, and the called party to priority x+1
- **A(x) -** Play an announcement (x.gsm) to the called party.
- **S(n) -** Hangup the call n seconds AFTER called party picks up.
- **d:** - This flag trumps the 'H' flag and intercepts any dtmf while waiting for the call to be answered and returns that value on the spot. This allows you to dial a 1-digit exit extension while waiting for the call to be answered - see also RetryDial
- **D(digits) -** After the called party answers, send digits as a DTMF stream, then connect the call to the originating channel.
- **L(x[:y][:z]) -** Limit the call to 'x' ms, warning when 'y' ms are left, repeated every 'z' ms) Only 'x' is required, 'y' and 'z' are optional. The following special variables are optional for limit calls: (pasted from app_dial.c)
  - **+ LIMIT_PLAYAUDIO_CALLER** - yes|no (default yes) - Play sounds to the caller.
  - **+ LIMIT_PLAYAUDIO_CALLEE** - yes|no - Play sounds to the callee.
  - **+ LIMIT_TIMEOUT_FILE** - File to play when time is up.
  - **+ LIMIT_CONNECT_FILE** - File to play when call begins.
  - **+ LIMIT_WARNING_FILE** - File to play as warning if 'y' is defined. If LIMIT_WARNING_FILE is not defined, then the default behaviour is to announce ('You have [XX minutes] YY seconds').
- **f** **-** forces callerid to be set as the extension of the line making/redirecting the outgoing call. For example, some PSTNs don't allow callerids from other extensions than the ones that are assigned to you.
- **w** **-** Allow the called user to start recording after pressing *1 or what defined in features.conf, requires Set(DYNAMIC_FEATURES=automon)
- **W** **-** Allow the calling user to start recording after pressing *1 or what defined in features.conf, requires Set(DYNAMIC_FEATURES=automon)

These options ring multiple extensions simultaneously

NOTE: If placing calls to mobile/proper number it may take 2-3 seconds until call is placed over Zaptel



Priority *:

Local/Proper/Mobile numbers to be dialed

Example: Provide a list of local/proper/mobile extensions that are to be dialled instantaneously when this extension is called. All extensions will for for a number of seconds provided in a 'Timeout' field. If none of the 'Priority' extensions answers the call, caller gets redirected to 'Last Destination' extension
Field Type: [0-9]

Timeout:

Ring time in seconds

Example: Setting '20' here will ring all 'Priority' extensions for 20 seconds. If none of the 'Priority' extensions answers the call, caller gets redirected to 'Last Destination' extension
Field Type: [0-9]

Last Destination:

The last destination to be dialled

Example: Extension number dialled if none of the 'Priority' extensions answers the call. In case that this 'Last Destination' does not answer the call, the call is redirected to 'Last Destination' voice box
Field Type: [0-9]

Dial Options:

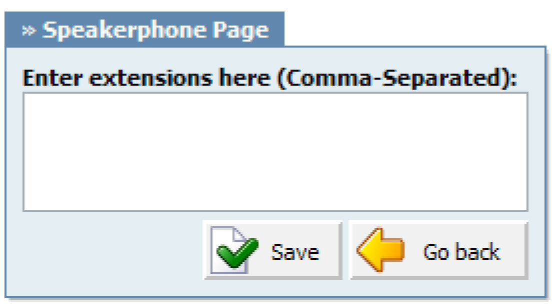
Additional call properties

Example: These options additionally customize the call properties such as allowing the called party to transfer the call, to play MOH music instead of the ringtone etc...
Field Type: [a-z]

- **t** - Allow the called user to transfer the call by hitting #
- **T** - Allow the calling user to transfer the call by hitting #
- **r** - Generate a ringing tone for the calling party, passing no audio from the called channel(s) until one answers. Use with care and don't insert this by default into all your dial statements as you are killing call progress information for the user. Really, you almost certainly do not want to use this. Asterisk will generate ring tones automatically where it is appropriate to do so. 'r' makes it go the next step and additionally generate ring tones where it is probably not appropriate to do so.
- **R** - Indicate ringing to the calling party when the called party indicates ringing, pass no audio until answered. This is available only if you are using kapejod's bristuff.
- **m** - Provide Music on Hold to the calling party until the called channel answers. This is mutually exclusive with option 'r', obviously. Use m(class) to specify a class for the music on hold.
- **o** - Restore the Asterisk v1.0 Caller ID behaviour (send the original caller's ID) in Asterisk v1.2 (default: send this extension's number)
- **j** - Asterisk 1.2 and later: Jump to priority n+101 if all of the requested channels were busy (just like behaviour in Asterisk 1.0.x)
- **M(x)** - Executes the macro (x) upon connect of the call (i.e. when the called party answers)
- **h** - Allow the callee to hang up by dialing *
- **H** - Allow the caller to hang up by dialing *
- **C** - Reset the CDR (Call Detail Record) for this call. This is like using the NoCDR command
- **P(x)** - Use the Privacy Manager, using x as the database (x is optional)
- **g** - When the called party hangs up, exit to execute more commands in the current context.
- **G(context^exten^pri) -** If the call is answered, transfer both parties to the specified priority; however

it seems the calling party is transferred to priority x, and the called party to priority x+1
- **A(x)** **-** Play an announcement (x.gsm) to the called party.
- **S(n)** **-** Hangup the call n seconds AFTER called party picks up.
- **d:** - This flag trumps the 'H' flag and intercepts any dtmf while waiting for the call to be answered and returns that value on the spot. This allows you to dial a 1-digit exit extension while waiting for the call to be answered - see also RetryDial
- **D(digits)** **-** After the called party answers, send digits as a DTMF stream, then connect the call to the originating channel.
- **L(x[:y][:z])** **-** Limit the call to 'x' ms, warning when 'y' ms are left, repeated every 'z' ms) Only 'x' is required, 'y' and 'z' are optional. The following special variables are optional for limit calls: (pasted from app_dial.c)
    - **+ LIMIT_PLAYAUDIO_CALLER** - yes|no (default yes) - Play sounds to the caller.
    - **+ LIMIT_PLAYAUDIO_CALLEE** - yes|no - Play sounds to the callee.
    - **+LIMIT_TIMEOUT_FILE** - File to play when time is up.
    - **+LIMIT_CONNECT_FILE** - File to play when call begins.
    - **+LIMIT_WARNING_FILE** - File to play as warning if 'y' is defined. If LIMIT_WARNING_FILE is not defined, then the default behaviour is to announce ('You have [XX minutes] YY seconds').
- **f** **-** forces callerid to be set as the extension of the line making/redirecting the outgoing call. For example, some PSTNs don't allow callerids from other extensions than the ones that are assigned to you.
- **w** **-** Allow the called user to start recording after pressing *1 or what defined in features.conf, requires Set(DYNAMIC_FEATURES=automon)
- **W** **-** Allow the calling user to start recording after pressing *1 or what defined in features.conf, requires Set(DYNAMIC_FEATURES=automon)

This option is used to transmit a message to multiple phones over their loudspeakers



Enter extensions here:

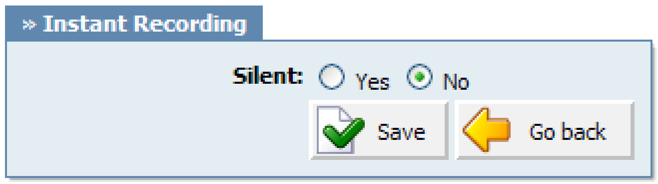
A group of extensions that will receive a call

Example: Setting this field to '1000,1001,1002' and dialing '*399' from this extension will send a call to loudspeakers of 1000, 1001 and 1002 UAD/Phone
Field Type: [0-9][,]

NOTE:
UAD/Phone by default has 10 seconds to auto answer
The call will be transferred to intercom if UAD/Phone supports this service, otherwise the call will fail

This option is used for instantaneous call recording. Anytime during conversation dial '*159' and the call will be recorded from the point you have dialled the number

NOTE: Access code for this service can be customized through 'Settings: Access Codes'



Silent:

Should parties in conversation be informed that call is being recorded

Example: With this option enabled, both parties in call will hear 'Recorded' message during the conversation
Field Type: Option buttons

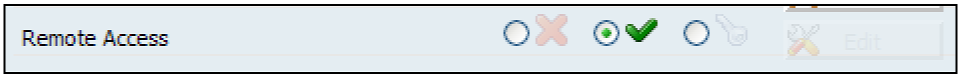
This option is used to delete recorded calls via 'Self Care: CDR: Advanced' options

This option is used to playback all extension recorded calls. To playback the files, login into 'Self Care', navigate to 'CDR', select a sound file you wish to hear and click on 'Play' button. The sound file will be downloaded to your desktop

NOTE: Sound files are marked with a round blue icon .

This option allow PBXware access from remote location.

For remote access IVR needs to be set with following options: 'IVR option 4 for example'='Remote access', 'Extensions'='Destination'. Once remote user enters IVR and presses digit '4', PBXware will ask for user extension and PIN number (both confirmed with '#' key). If provided extension and PIN numbers are valid, user will hear a dial tone and will be able to make a call.

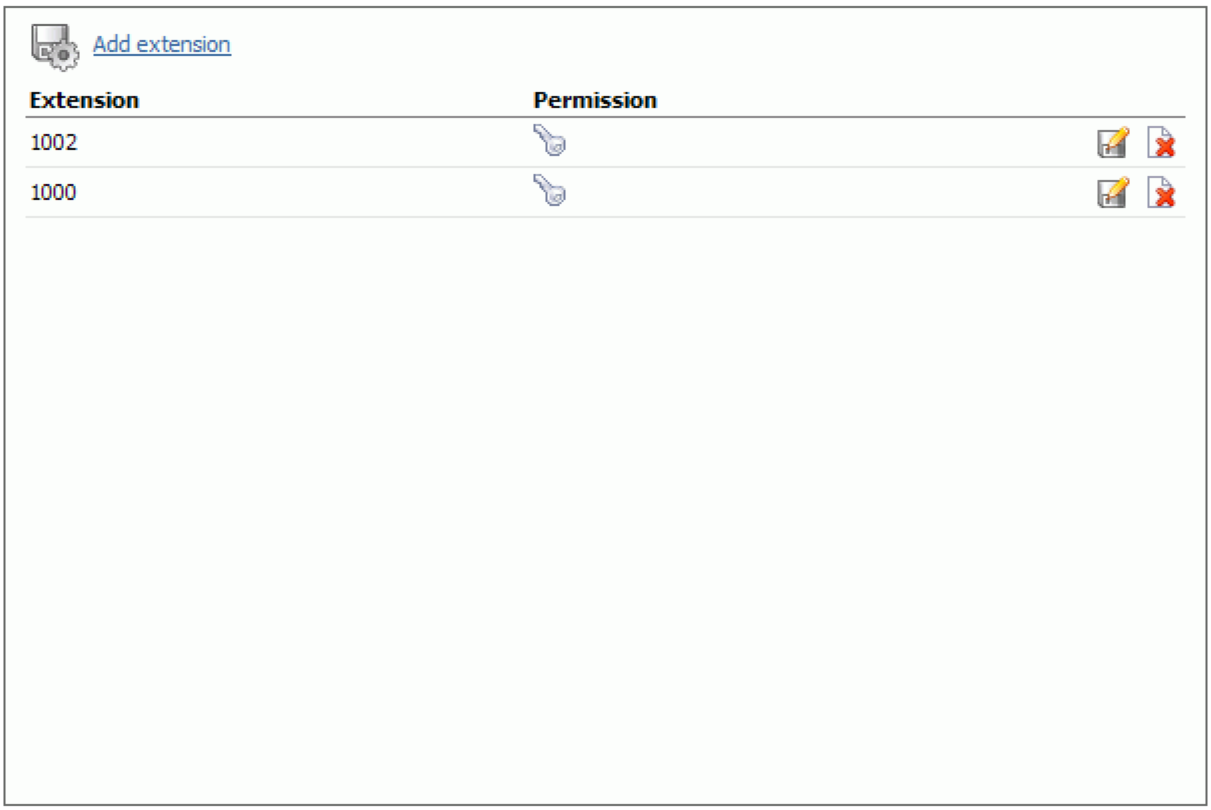


4.1.1.3.11.4 Call Monitoring

This option allows user to monitor conversations of other PBXware extensions in real time.

For example, extensions 1000 and 1001 are in conversation. Extension 1005 dials '*199+1000'. From that moment, active call and all other calls made by extension 1000 will be heard by extension 1005 until extension 1005 hangs up

Window bellow displays all monitored extensions by extension number and monitoring permission



Extension:
Monitored UAD/Phone extension number
Example: 1002
Field Type: Display

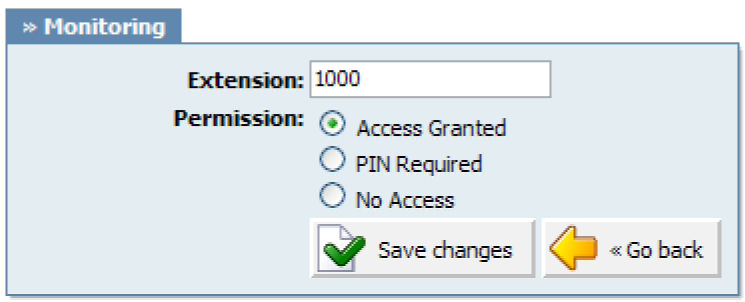
Permission:

Permissions for monitored UAD/Phone
Example: PIN
Field Type: Display


Edit monitored UAD/Phone configuration
Example: Click to edit monitored UAD/Phone configuration
Field Type: Button


Delete monitored UAD/Phone configuration
Example: Click to delete monitored UAD/Phone configuration
Field Type: Button

Clicking on 'Add extension' button will add new extension to monitoring list



Extension:

Extension number that is to be monitored.

Example: Set '1000' here to monitor extension with that network number. Set 'ALL' here to monitor all PBXware extensions
Field Type: [0-9] [ALL]

Permission:

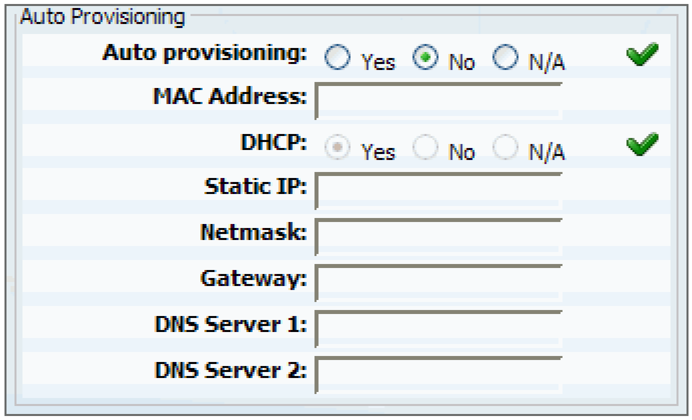
Permissions for accessing monitored extension

Example:
- Access Granted - Monitor by dialling '*199+$EXTENSION' only (without providing PIN number)
- PIN Required - Monitor by dialling '*199+$EXTENSION' and providing a PIN number
- No Access - Do not monitor this extension but leave its configuration in 'Monitoring' window

Field Type: Select box

4.1.1.3.12 Auto Provisioning

These options are used for UAD/Phone auto provisioning. configuration file is created on PBXware TFTP server from where UAD/Phone picks it up



Auto Provisioning:

Enable auto provisioning service for this extension

Example: Select 'Yes' to enable auto provisioning and provide more details in the fields bellow
Field Type: Option Buttons

MAC Address(Media Access Control):

UAD/Phone MAC address

Example: Every UAD/Phone has a unique MAC address. It is a 48-bit hexadecimal number, 12 characters long and usually printed on UAD's/Phone's back. Provide this MAC address here e.g. 000E08DE31EB
Field Type: [a-z] [0-9]

DHCP (Dynamic Hosts Configuration Protocol):

Select whether UAD/Phone is on DHCP or static IP address

Example: By default all UADs/Phones should be on dynamic IP addresses. In this case select 'DHCP' here. If UAD/Phone is in static IP address, select 'Static' here and provide additional network information in the fields bellow
Field Type: Option buttons

Static IP:

Static UAD/Phone IP address

Example: Set UAD/Phone static IP address here e.g. 192.168.8.221
Field Type: [0-9][.]

Netmask:

Netmask applied for UAD/Phone static IP address

Example: If UAD/Phone static IP address is 192.168.8.221, netmask set here is 255.255.255.0
Field Type: [0-9][.]

Gateway:

UAD/Phone gateway IP address

Example: If UAD/Phone static IP address is 192.168.8.221, gateway set here by default would be 192.168.8.1
Field Type: [0-9][.]

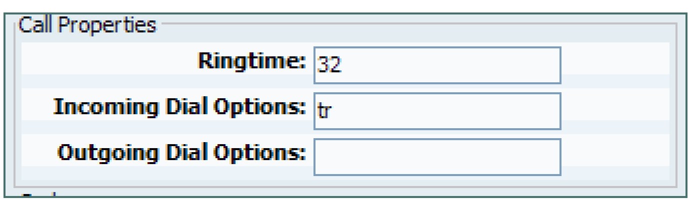
DNS Server1 and Server2(Domain Name Server):

DNS Server IP address

Example: LAN DNS IP address (usually the same IP address as your gateway e.g. 192.168.8.1)
Field Type: [0-9][.]

4.1.1.3.13 Call Properties

These options fine tune advanced incoming/outgoing call settings



Ringtime:

UAD/Phone ring time in seconds

Example: If '32' is set here, UAD/Phone will ring for 32 seconds before the call is considered unanswered
Field Type: [0-9]

Incoming Dial Options:

Advanced dial options for all incoming calls

Example: This option fine tunes incoming call properties. By providing 't' here, receiver will be able to transfer caller to other extensions. For a list of all advanced options see bellow
Field Type: [a-z]

Outgoing Dial Options:

Advanced dial options for all outgoing calls

Example: This option fine tunes outgoing call properties. By providing 'T' here, caller will be able to transfer receiver to other extensions. For a list of all advanced options see bellow
Field Type: [a-z]

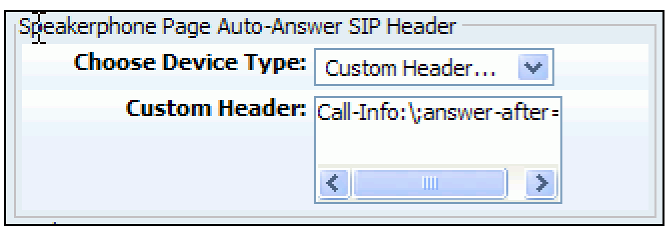
Dial Options:

- **t** **-** Allow the called user to transfer the call by hitting #
- **T** **-** Allow the calling user to transfer the call by hitting #
- **r** **-** Generate a ringing tone for the calling party, passing no audio from the called channel(s) until one answers. Use with care and don't insert this by default into all your dial statements as you are killing call progress information for the user. Really, you almost certainly do not want to use this. Asterisk will generate ring tones automatically where it is appropriate to do so. 'r' makes it go the next step and additionally generate ring tones where it is probably not appropriate to do so.
- **R** **-** Indicate ringing to the calling party when the called party indicates ringing, pass no audio until answered. This is available only if you are using kapejod's bristuff.
- **m** **-** Provide Music on Hold to the calling party until the called channel answers. This is mutually exclusive with option 'r', obviously. Use m(class) to specify a class for the music on hold.
- **o** **-** Restore the Asterisk v1.0 Caller ID behaviour (send the original caller's ID) in Asterisk v1.2 (default: send this extension's number)
- **j** **-** Asterisk 1.2 and later: Jump to priority n+101 if all of the requested channels were busy (just like behaviour in Asterisk 1.0.x)
- **M(x)** **-** Executes the macro (x) upon connect of the call (i.e. when the called party answers)
- **h** **-** Allow the callee to hang up by dialing *
- **H** **-** Allow the caller to hang up by dialing *
- **C** **-** Reset the CDR (Call Detail Record) for this call. This is like using the NoCDR command
- **P(x)** **-** Use the Privacy Manager, using x as the database (x is optional)
- **g** **-** When the called party hangs up, exit to execute more commands in the current context.
- **G(context^exten^pri) -** If the call is answered, transfer both parties to the specified priority; however it seems the calling party is transferred to priority x, and the called party to priority x+1

- **A(x)** - Play an announcement (x.gsm) to the called party.
- **S(n)** - Hangup the call n seconds AFTER called party picks up.
- **d:** - This flag trumps the 'H' flag and intercepts any dtmf while waiting for the call to be answered and returns that value on the spot. This allows you to dial a 1-digit exit extension while waiting for the call to be answered - see also RetryDial
- **D(digits)** - After the called party answers, send digits as a DTMF stream, then connect the call to the originating channel.
- **L(x[:y][:z])** - Limit the call to 'x' ms, warning when 'y' ms are left, repeated every 'z' ms) Only 'x' is required, 'y' and 'z' are optional. The following special variables are optional for limit calls: (pasted from app_dial.c)
    - **+ LIMIT_PLAYAUDIO_CALLER** - yes|no (default yes) - Play sounds to the caller.
    - **+ LIMIT_PLAYAUDIO_CALLEE** - yes|no - Play sounds to the callee.
    - **+LIMIT_TIMEOUT_FILE** - File to play when time is up.
    - **+LIMIT_CONNECT_FILE** - File to play when call begins.
    - **+LIMIT_WARNING_FILE** - File to play as warning if 'y' is defined. If LIMIT_WARNING_FILE is not defined, then the default behaviour is to announce ('You have [XX minutes] YY seconds').
- **f** - forces callerid to be set as the extension of the line making/redirecting the outgoing call. For example, some PSTNs don't allow callerids from other extensions than the ones that are assigned to you.
- **w** - Allow the called user to start recording after pressing *1 or what defined in features.conf, requires Set(DYNAMIC_FEATURES=automon)
- **W** - Allow the calling user to start recording after pressing *1 or what defined in features.conf, requires Set(DYNAMIC_FEATURES=automon)

4.1.1.3.14 Speakerphone Page Auto-Answer SIP Header

These options are used to transmit a message to multiple phones over their loudspeakers



Choose Device Type:

Set predefined UAD/Phone type for this extension

Example: By selecting a device type here (for Sipura phones select 'Sipura' here), custom SIP page header will be assigned to this extension every time paging is executed
Field Type: Select box

Custom Header:

Custom SIP page header

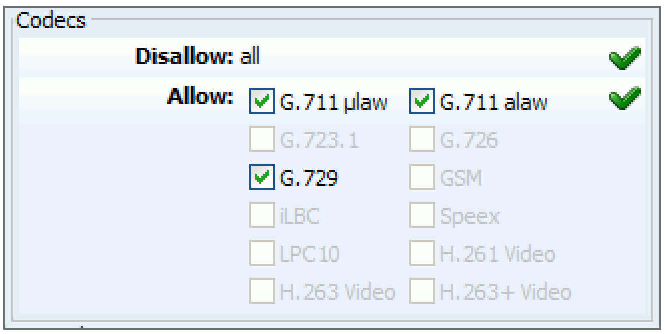
Example: If UAD/Phone is not listed under 'Choose Device Type' option, you may set a custom SIP header

string here. Default value for all other UADs/Phones is 'Call-Info:\;answer-after=0'
Field Type: [a-z][0-9]

4.1.1.3.15 Codecs

Codecs are used to convert analog to digital voice signals and vice versa. These options set preferred codecs used by the extension.

NOTE: If some of the codecs are disabled (cannot be selected), navigate to 'Settings: Servers: Edit: Default Codecs' and enable them under the 'Local' group



Disallow:

This field is very unique indeed. In order to work properly, this option must always be set to 'disallow all'

Allow:

Codecs UAD/Phone is allowed to use

Example: Codecs vary in the sound quality/bandwidth they use (see bellow for more info). Select a box next to a codec name to allow UAD/Phone to use that codec
Field Type: Check box

Codec bandwidth usage:

- **ITU G.711 ulaw** - 64 Kbps, sample-based, used in US
- **ITU G.711 alaw** - 64 Kbps, sample-based, used in Europe
- **ITU G.723.1** - (5.3/6.3 Kbps, 30ms frame size
- **ITU G.726** - 16/24/32/40 Kbps
- **ITU G.729** - 8 Kbps, 10ms frame size
- **GSM** - 13 Kbps (full rate), 20ms frame size
- **iLBC** - 15Kbps,20ms frame size: 13.3 Kbps, 30ms frame size

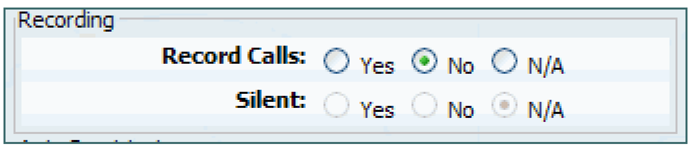
- **Speex** - 2.15 to 44.2 Kbps
- **LPC10** - 2.5 Kbps
- **H.261 Video** - Used over ISDN lines with resolution of 352x288
- **H.263 Video** - Low-bit rate encoding solution for video conferencing
- **H.263+ Video** - Extension of H.263 that provides additional features that improve compression over packet switched networks.

4.1.1.3.16 Recording

This group of options is used to record all incoming/outgoing calls made by the extension

NOTE:
- Laws in some countries may require notifying the parties that their call is being recorded.

- Recorded calls, marked with icon , can be access from 'Self Care Interface' or 'Reports: CDR' PBXware' menu.
- Call are recorded in audio format set under 'Settings: Servers: Recordings Format'.



Record Calls:

Enable call recording service

Example: Select 'Yes' to enable the service. All incoming/outgoing calls will be recorded. If using call recording with many extensions, check server disk space from time to time. Please see below for bit rates table
Field Type: Options buttons

Silent:

Set weather call recordings should be announced to parties in conversation

Example: If Silent=No, calling parties will hear 'Recorded' or 'This call is recorded' message before their conversation starts
Field Type: Options buttons

Disk Space Used By Call Recording:

With continuously tone 60 seconds:

- wav49 = 84.5kb
- wav = 833.0kb
- gsm = 85.0kb

With continuously silent tone (without sound) 60 seconds:

- wav49 = 84.0kb
- wav = 827.0kb
- gsm = 84.0kb

4.1.1.3.17 Presence

Some UADs/Phones support hints/presence feature. This feature identifies the status of other UADs/ Phones on the network. Properly identifying UAD/Phone status results in batter handling of calls. For example, if extension is busy, caller will hear busy and not the unavailable dial sound

Presence/Hints supported UADs:

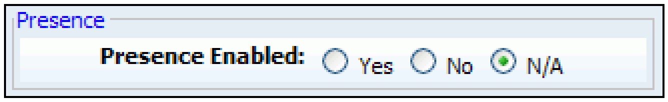
- Snom 190(Firmware >= 3.60s), 320/360(Firmware >= 4.1)
- Polycom IP30x/IP50x/IP600
- Xten EyeBeam
- Grandstream GXP2000 (Firmware >= 1.0.1.13)
- Aastra 480i
- Aastra 9133i

CLI:

Presence/Hints can be checked by command line as well. From PBXware console type '/home/ servers/pbxware/sh/asterisk -rvvvvvvvvvv' and type 'show hints' to view the presence information.

```
> show hints
   -= Registered Asterisk Dial Plan Hints =-
  1009          : SIP/1009        State:Idle          Watchers  0
  2001          : SIP/2001        State:Idle          Watchers  0
  1020          : SIP/1020        State:InUse         Watchers  0
  1016          : SIP/1016        State:Unavailable   Watchers  0
  1008          : SIP/1008        State:Idle          Watchers  0
  1006          : SIP/1006        State:Unavailable   Watchers  0
  1000          : SIP/1000        State:Ringing       Watchers  0
  1003          : SIP/1003        State:Unavailable   Watchers  0
  1030          : SIP/1030        State:Unavailable   Watchers  0
  1234          : IAX2/1234       State:Unavailable   Watchers  0
  7777          : IAX2/7777       State:Idle          Watchers  0
  1017          : IAX2/1017       State:Unavailable   Watchers  0
```

- 12 hints registered



Presence Enabled:

Enable presence/hints service

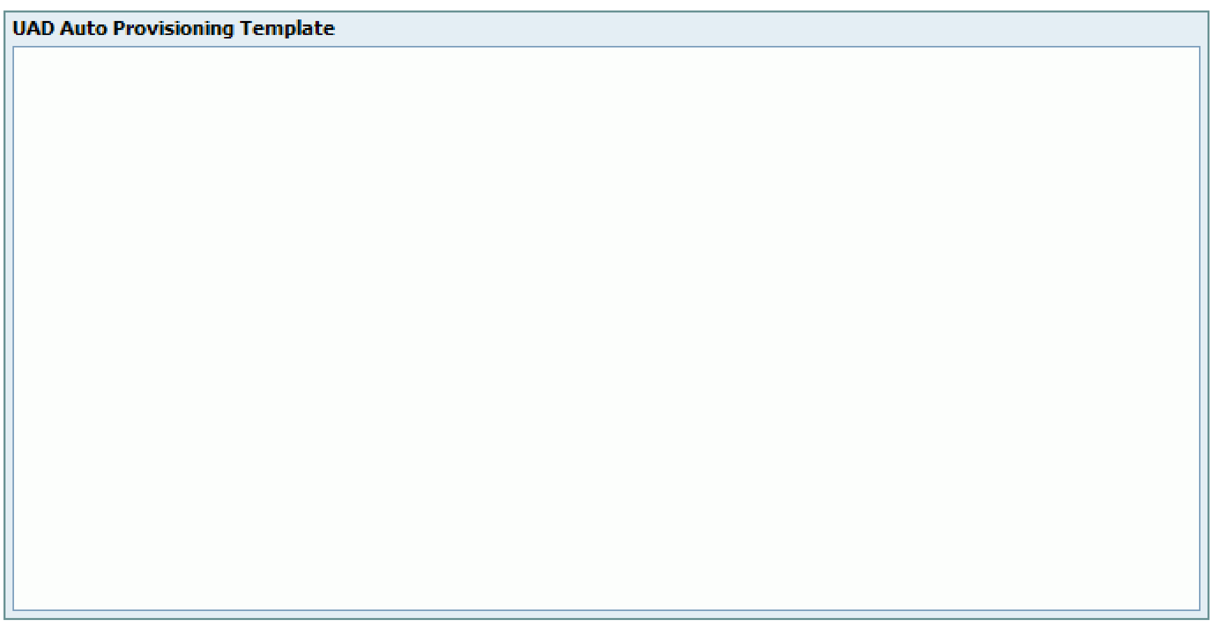
Example: Select 'Yes' to enable presence support
Field Type: Option buttons

4.1.1.3.18 User Agent Auto Provisioning Template

This option allows adding of additional settings to auto-provisioning template.

Auto-provisioning settings are generally defined in the 'Settings: UAD' and are custom set for each device.

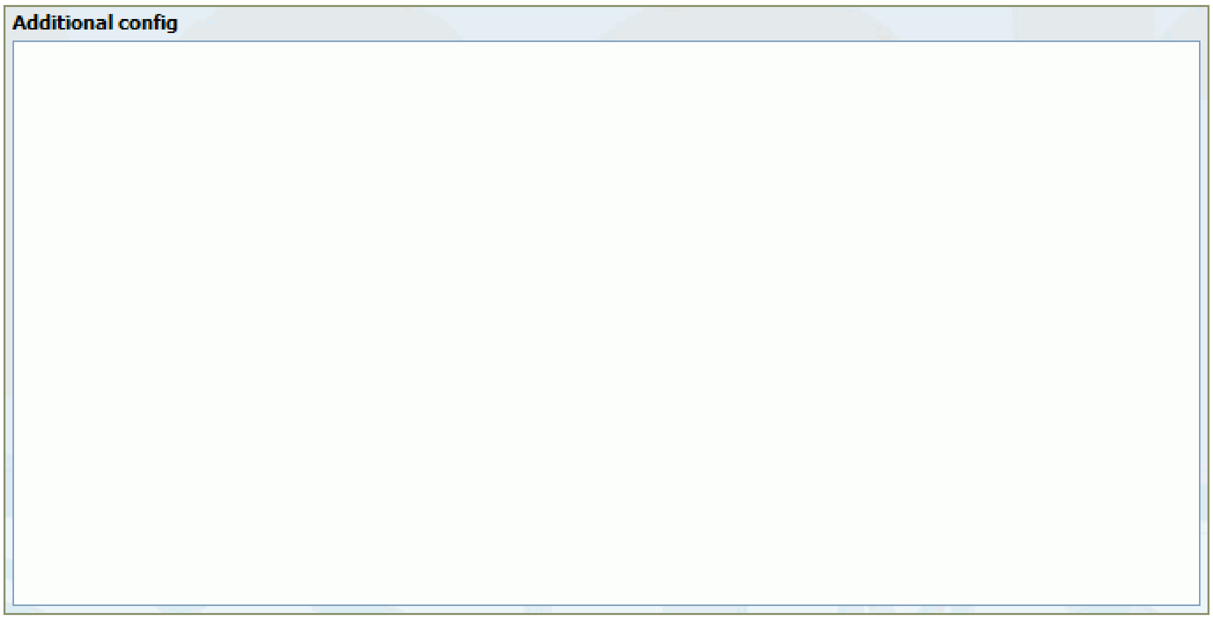
NOTE: Unless absolutely sure, do not change or add to this template.



4.1.1.3.19 Additional Config

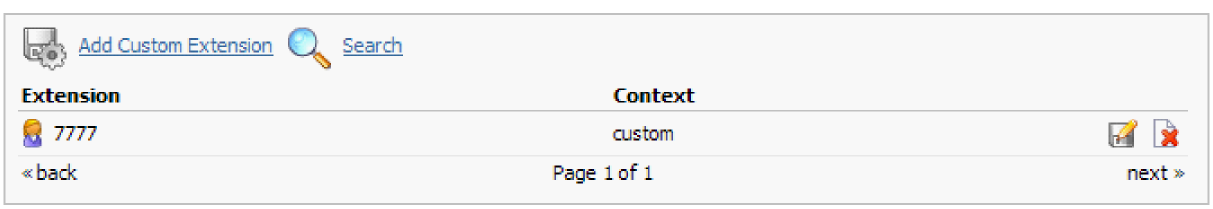
This option is used for providing additional config parameters for SIP, IAX and MGCP configuration files. Values provided here will be written into these configuration files.

NOTE: Unless absolutely sure, do not change or add to this template.



4.2 Custom

Custom extensions are used for executing custom PBXware commands/scripts. This screen lists all PBXware custom extensions with the following details:



Extension:

Custom extension network number
Example: Dial this number to execute a custom PBXware script (e.g. 7777)
Field Type: Display

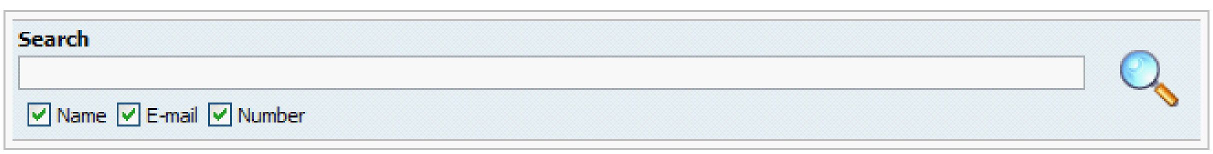
Context:
Context custom extension belongs to
Example: custom
Field Type: Display


Edit custom extension configuration
Example: Click to edit custom extension configuration
Field Type: Button


Delete custom extension configuration
Example: Click to delete a custom extension from the system
Field Type: Button

4.2.1 Search

Search bar filters custom extensions by name, e-mail and number



Search:
Search phrase
Example: Provide a search phrase here and hit enter to filter the records
Field Type: [a-z][0-9]

Name:
Should search filter be applied to names UADs are registered to
Example: Check the box to search the names
Field Type: Check box

E-mail:
Should search filter be applied to email addressed associate with the UADs
Example: Check the box to search the email addresses
Field Type: Check box

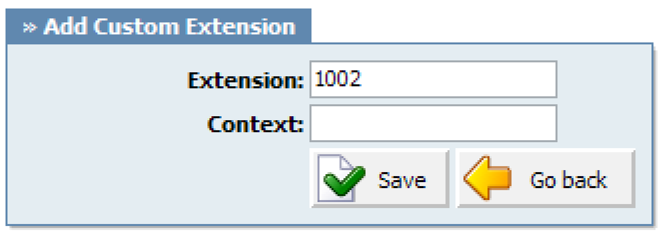
Number:
Should search filter be applied to extension numbers

Example: Check the box to search extension numbers
Field Type: Check box

4.2.2 Add/Edit Custom Extension

To add new custom extension click on 'Add custom extension' button

NOTE: Custom extensions do not belong to 'default' system context. Their actions are set under 'Settings: Conf Files: Extensions' configuration file



Extension:

System extension number

Example: Setting '1002' here will create new system extension with the same network number. Dial this number to execute custom script set under 'Settings: Conf Files: Extensions'
Field Type: [0-9]

Context:

Every system extension belongs to a certain system context. Context may be described as a collection/group of extensions. Default context used by the PBXware is 'default' and must not be used by Custom extensions

Example: Set this field to 'test' for example
Field Type: [a-z] [0-9]

Extensions.conf sample:

Navigate to 'Settings: Conf Files: Extensions' and add following lines to the bottom of the file

```
[test]
exten => 1002,1,Answer()
exten => 1002,2,Playback(hello-world)
exten => 1002,3,Hangup()
```

Click 'Save' button.
Reload PBXware and dial 1002 extension.
'Hello world' will be played from the other end :)

4.3 Ring Groups

Ring Groups are used to group a number of UADs/Phones into one network destination. Each Ring Group is assigned a network number which once dialed rings all extensions assigned to group.

Add Ring Group

Ring Group	Extension	Destinations	Last Destination
Accounts	1111	1001, 1002, 1003, 1004 ...	1010
Billing	2222	1001, 1002, 1003, 1004	-
Customer Services	3333	1001, 1002, 1003, 1004	-

Ring Group:
Ring group name
Example: Accounts
Field Type: Display

Extension:
Ring group extension number
Example: Once user dials this number all destinations assigned to ring group will ring (e.g. 1111)
Field Type: Display

Destinations:
Extension numbers assigned to a ring group
Example: Once a ring group number is dialed all destinations set here will ring at the same time (e.g. 1001, 1002, 1003...)
Field Type: Display

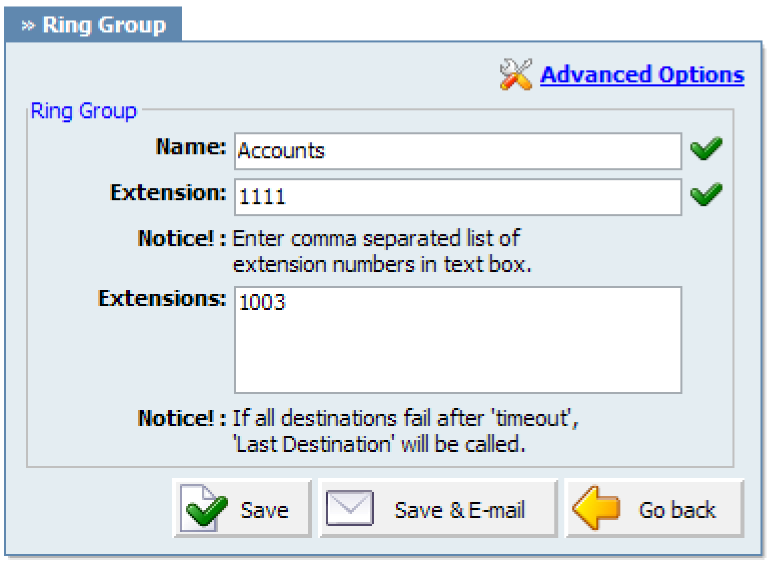
Last Destination:
Last destination to be called if none of the destination extensions answers the call
Example: 1010
Field Type: Display

Edit ring group configuration
Example: Click to edit ring group configuration
Field Type: Button

Delete ring group from the system
Example: Click to delete a ring group from the system
Field Type: Button

4.3.1 Add/Edit Ring Group

Clicking on 'Add Ring Group' or 'Edit' button will open following ring group options:



Name:

Unique Ring group name

Example: Set 'Accounts' here to create the same ring group
Field Type: [a-z][0-9]

Extension:

Unique network number associated with the Ring group

Example: When this number is dialed, all extensions associated with it will ring at the same time
Field Type: [0-9]

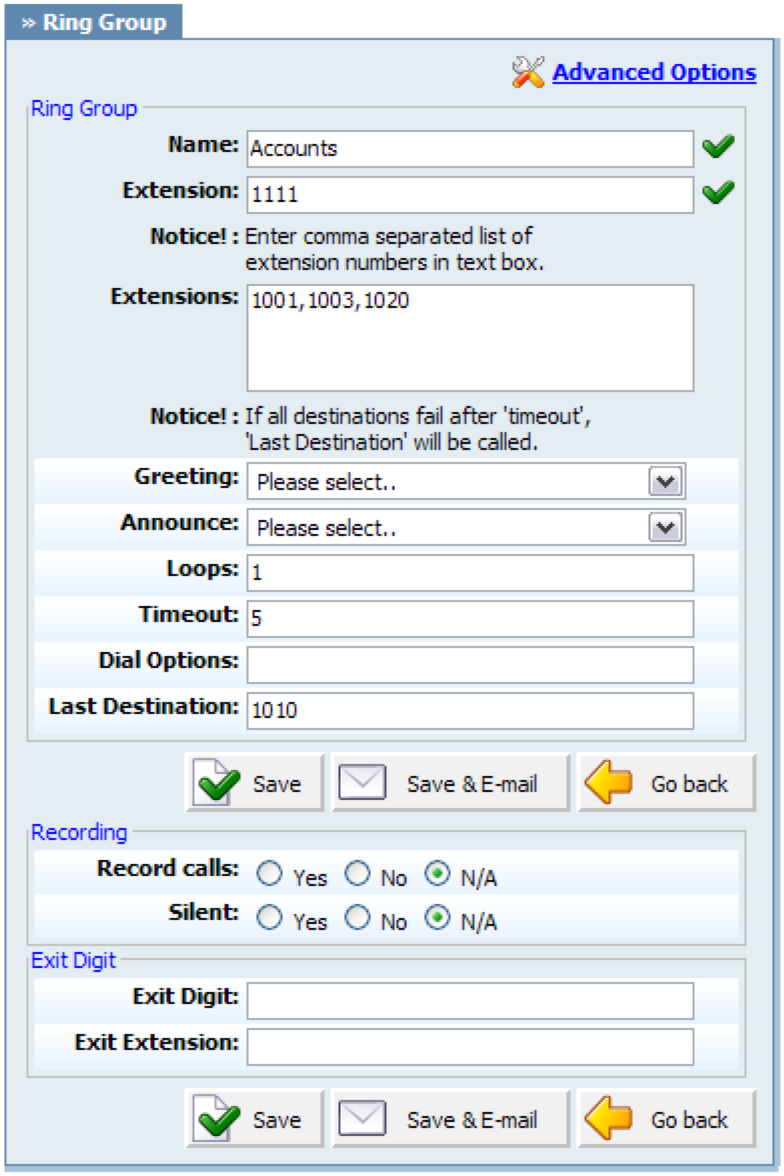
Extensions:

System extensions associated with the ring group

Example: Provide an extension list separated by commas here (e.g. 1001,1002,1003...). When ring group 'Extension' number is dialed, all extensions set here will ring at the same time. NOTE: If all destinations fail after 'timeout',
'Last Destination' will be called.
Field Type: [0-9]

4.3.2 Advanced Options

These options fine-tune ring group settings with additional options



Greeting:

Greeting sound file played to callers when Ring group is dialed

Example: Select 'greeting-default-attendant' for example. Any user that calls this ring group will hear this sound file played to them before all ring group extensions are dialed
Field Type: Select box

Announce:

Sound file played to caller if his call does not get answered by any of the ring group extensions. NOTE:

Sound file must have 'announce-' name prefix (e.g. 'announce-unavailable')

Example: John dials ring group 1000, but nobody answered his call. Sound file selected here will be played to John and then his call will be transferred to 'Last Destination' extension.
Field Type: Select box

Loops:

How many times to dial all extensions again if nobody answers

Example: John dials Ring group 1000, but nobody answered his call. If this option is set to '2', all extensions will be dialed one more time before transferring his call to 'Last Destination'
Field Type: [0-9]

Timeout:

How many seconds will all ring group extension ring before the call is considered unanswered

Example: This option is set to 20. John dials ring group 1000. All Extensions will ring for 20 seconds before timeout occurs. Depending on whether 'Loop' option is set, all extension will be rung again, or John will be transferred to 'Last Destination'
Field Type: [0-9]

Dial Options:

Additional call options assigned to a ring group

Example: To play music to ring group callers set this field to 'm($CLASS)', where m = MOH class e.g. m('default'). Please check details on the bottom
Field Type: [a-z]

Last Destination:

Last destination to be dialed if none of the ring group extensions answers the call

Example: John dials Ring group 1000, but nobody answered his call. Sound file selected under 'Announce' is played to John and his call is be transferred to extension number set here.
Field Type: [0-9]

Record Calls:

Enable call recording service

Example: Select 'Yes' to enable the service. All incoming/outgoing calls will be recorded. If using call recording with many extensions, check server disk space from time to time. Please see below for bit rates table.
Field Type: Options buttons

Silent:

Set weather call recordings should be announced to parties in conversation.

Example: If Silent=No, calling parties will hear 'Recorded' or 'This call is recorded' message before their conversation starts
Field Type: Options buttons

Exit Digit:

Exit digit that transfers the call to 'Exit Destination'

Example: John dials ring group 1000. While all extensions are ringing John presses 'Exit digit' set here (e.g. 9) and his call is transferred to 'Exit Destination'.
Field Type: [0-9]

Exit Extension:

PBXware extension to which the call is transferred once user dials the 'Exit Digit'

Example: John dials ring group 1000. While all extensions are ringing John presses the 'Exit Digit' and his call is transferred to 'Exit Destination' provided here (e.g. 2001)
Field Type: [0-9]

Disk Space Used By Call Recording:

With continuously tone 60 seconds:

- wav49 = 84.5kb
- wav = 833.0kb
- gsm = 85.0kb

With continuously silent tone (without sound) 60 seconds:

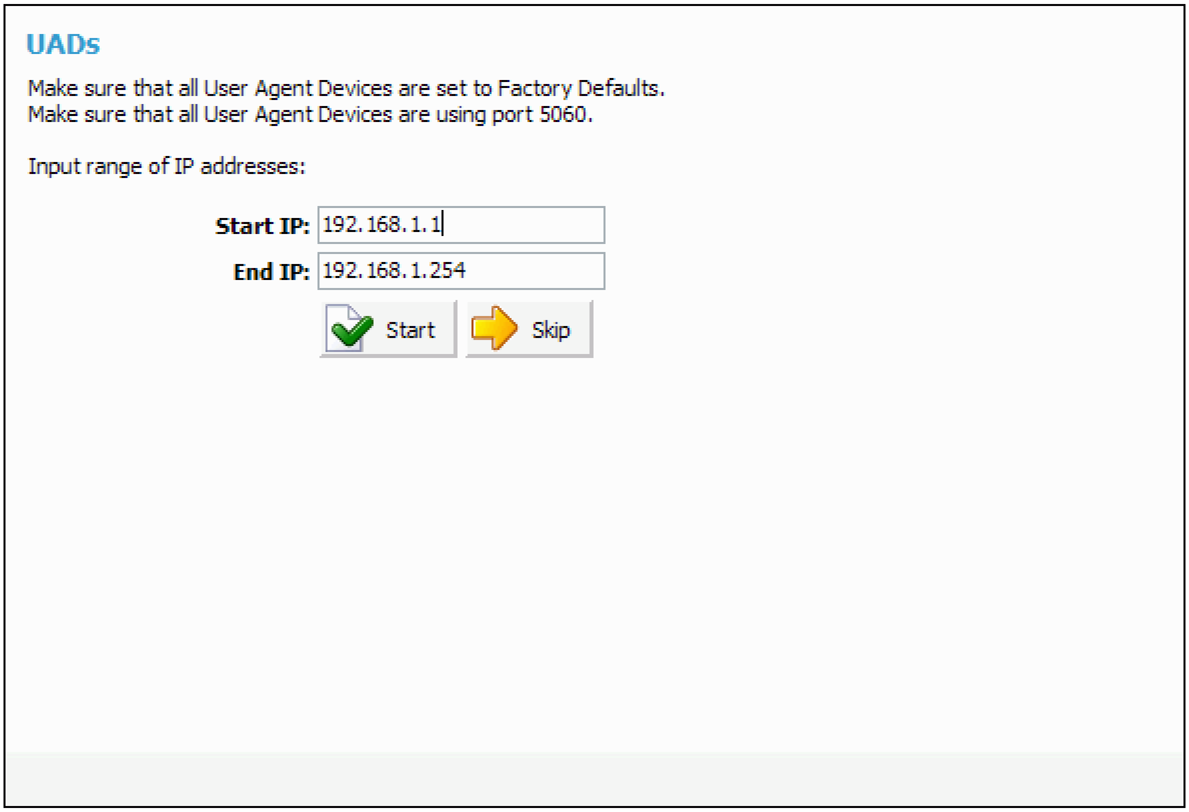
- wav49 = 84.0kb
- wav = 827.0kb
- gsm = 84.0kb

Dial Options:

- **t -** Allow the called user to transfer the call by hitting #
- **T -** Allow the calling user to transfer the call by hitting #
- **r -** Generate a ringing tone for the calling party, passing no audio from the called channel(s) until one answers. Use with care and don't insert this by default into all your dial statements as you are killing call progress information for the user. Really, you almost certainly do not want to use this. Asterisk will generate ring tones automatically where it is appropriate to do so. 'r' makes it go the next step and additionally generate ring tones where it is probably not appropriate to do so.
- **R -** Indicate ringing to the calling party when the called party indicates ringing, pass no audio until answered. This is available only if you are using kapejod's bristuff.
- **m -** Provide Music on Hold to the calling party until the called channel answers. This is mutually exclusive with option 'r', obviously. Use m(class) to specify a class for the music on hold.
- **o -** Restore the Asterisk v1.0 Caller ID behaviour (send the original caller's ID) in Asterisk v1.2 (default: send this extension's number)
- **j -** Asterisk 1.2 and later: Jump to priority n+101 if all of the requested channels were busy (just like behaviour in Asterisk 1.0.x)
- **M(x) -** Executes the macro (x) upon connect of the call (i.e. when the called party answers)
- **h -** Allow the callee to hang up by dialing *
- **H -** Allow the caller to hang up by dialing *
- **C -** Reset the CDR (Call Detail Record) for this call. This is like using the NoCDR command
- **P(x) -** Use the Privacy Manager, using x as the database (x is optional)
- **g -** When the called party hangs up, exit to execute more commands in the current context.
- **G(context^exten^pri) -** If the call is answered, transfer both parties to the specified priority; however it seems the calling party is transferred to priority x, and the called party to priority x+1
- **A(x) -** Play an announcement (x.gsm) to the called party.
- **S(n) -** Hangup the call n seconds AFTER called party picks up.
- **d:** - This flag trumps the 'H' flag and intercepts any dtmf while waiting for the call to be answered and returns that value on the spot. This allows you to dial a 1-digit exit extension while waiting for the call to be answered - see also RetryDial
- **D(digits) -** After the called party answers, send digits as a DTMF stream, then connect the call to the originating channel.
- **L(x[:y][:z]) -** Limit the call to 'x' ms, warning when 'y' ms are left, repeated every 'z' ms) Only 'x' is required, 'y' and 'z' are optional. The following special variables are optional for limit calls: (pasted from app_dial.c)
    - **+ LIMIT_PLAYAUDIO_CALLER** - yes|no (default yes) - Play sounds to the caller.
    - **+ LIMIT_PLAYAUDIO_CALLEE** - yes|no - Play sounds to the callee.
    - **+LIMIT_TIMEOUT_FILE** - File to play when time is up.
    - **+LIMIT_CONNECT_FILE** - File to play when call begins.
    - **+LIMIT_WARNING_FILE** - File to play as warning if 'y' is defined. If LIMIT_WARNING_FILE is not defined, then the default behaviour is to announce ('You have [XX minutes] YY seconds').
- **f -** forces callerid to be set as the extension of the line making/redirecting the outgoing call. For example, some PSTNs don't allow callerids from other extensions than the ones that are assigned to you.
- **w -** Allow the called user to start recording after pressing *1 or what defined in features.conf, requires Set(DYNAMIC_FEATURES=automon)
- **W -** Allow the calling user to start recording after pressing *1 or what defined in features.conf, requires Set(DYNAMIC_FEATURES=automon)

4.4 Find Phones

Find Phones is used to search and configure all available UADs/Phones on the local network. NOTE: Please make sure that all UADs/Phones are reset to factory defaults and using port 5060 before performing the search operation.



Start IP:
IP address from which search is started
Example: 192.168.1.1
Field Type: IP Address

End IP:
IP address with which search is ended
Example: 192.168.1.254
Field Type: IP Address

Start:
Start UADs network search
Example: Click to start
Field Type: Button

Skip:
Skip UADs network search
Example: Click to skip the search step
Field Type: Button

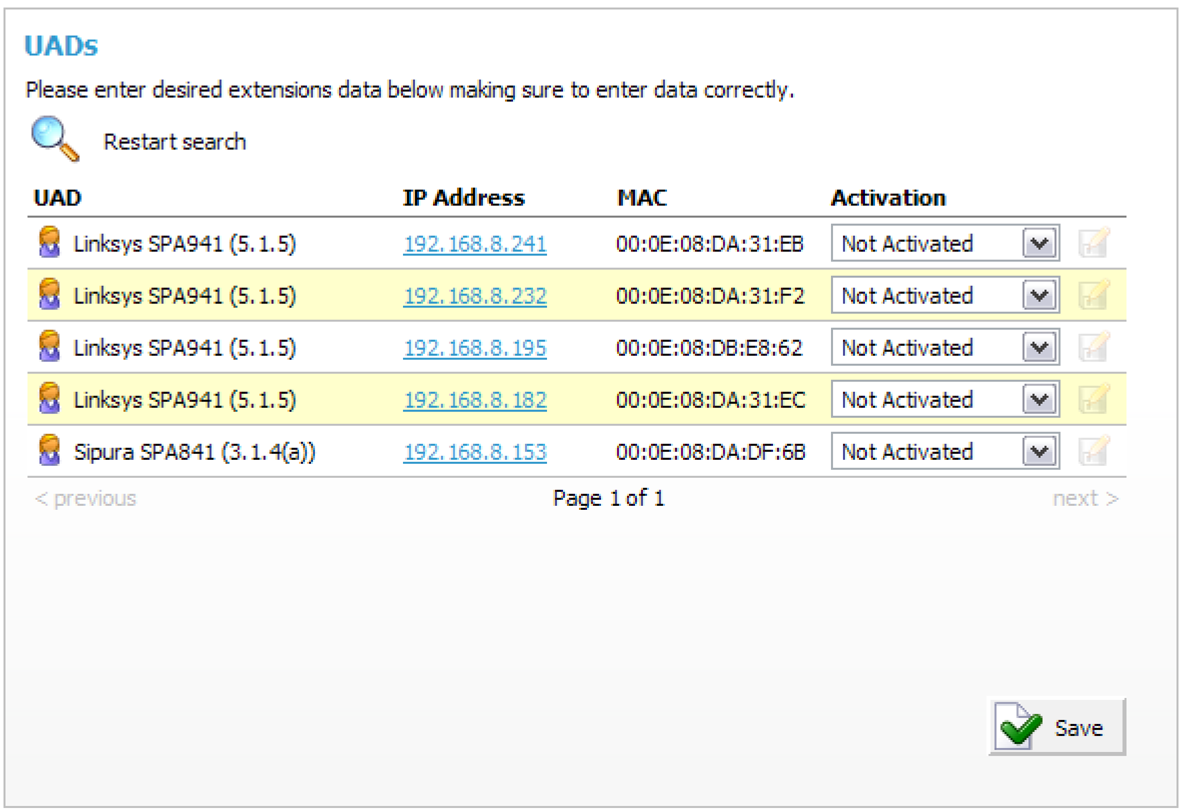
4.4.1 UADS

In this chapter we will cover:
- Find UADs 108
- Configure UADs 109

4.4.1.1 Found UADs

This screen displays all UADs/Phones found on the local network and displays them with the following details:

NOTE: In order to restart the search, please click on the 'Restart search' button



UAD:

UAD/Phone model detected
Example: Linksys SPA941 (5.1.5)
Field Type: Display

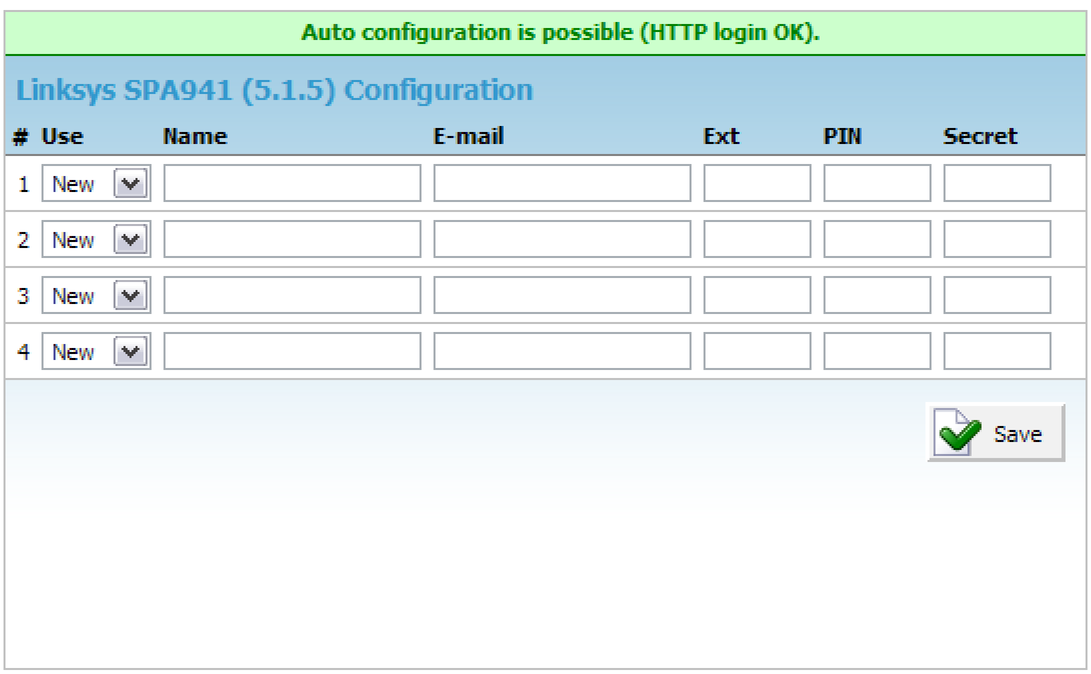
IP Address:
UAD/Phone IP address on the network
Example: 192.168.8.241. If this ip is clicked, UAD/Phone web interface will open in new browser window
Field Type: Display

MAC:
UAD/Phone MAC address
Example: 00:0E:0E:DE:3E:EE
Field Type: Display

Activation:
Set the way UAD/Phone is to be configured
Example:
Not Activated - Skip extension
Activated - Create extension in interface only and do nothing else
Auto Provisioning - Create extension in interface and configuration file in TFTP directory (Requires device reboot)
Auto Configuration - Create extension in interface, configuration file in TFTP, go into UAD/Phone web interface, configure details there and reboot the phone
Field Type: Select box

4.4.1.2 Configure UADs

This screen displays the options for configuring of the UAD/Phone lines:



Use:

Select whether new phone line is to be created or existing system extension details are to be applied

Example: If you select 'New' provide all necessary details for the extension or select one of the system extensions to configure phone line with its details
Field Type: Select box

Name:

Full name of the person using the Extension. This name i sent in a Caller ID information

Example: Setting 'Joanna Cox' here will show this name on other UAD/Phone display when the call is made
Field Type: [a-z][0-9]

E-mail:

Email address associated with the extension and used for various system notifications

Example: Setting 'joanna@domain.com' here will transfer all voicemail notifications, Extension PIN and other details to this email
Field Type: [a-z][0-9]

Ext:

System extension number

Example: Setting '1008' here will create new system extension with the same network number. By default, this field is automatically populated, but can be changed to any extension number.
Field Type: [0-9]

PIN (Personal Identification Number):

Four digit number used for account authorization. NOTE: This number must always be four (4) digits long

Example: If PIN for this extension is set to '8474', provide it when asked for it by the PBXware when checking your voice inbox or other enhanced services
Field Type: [0-9]

Secret:

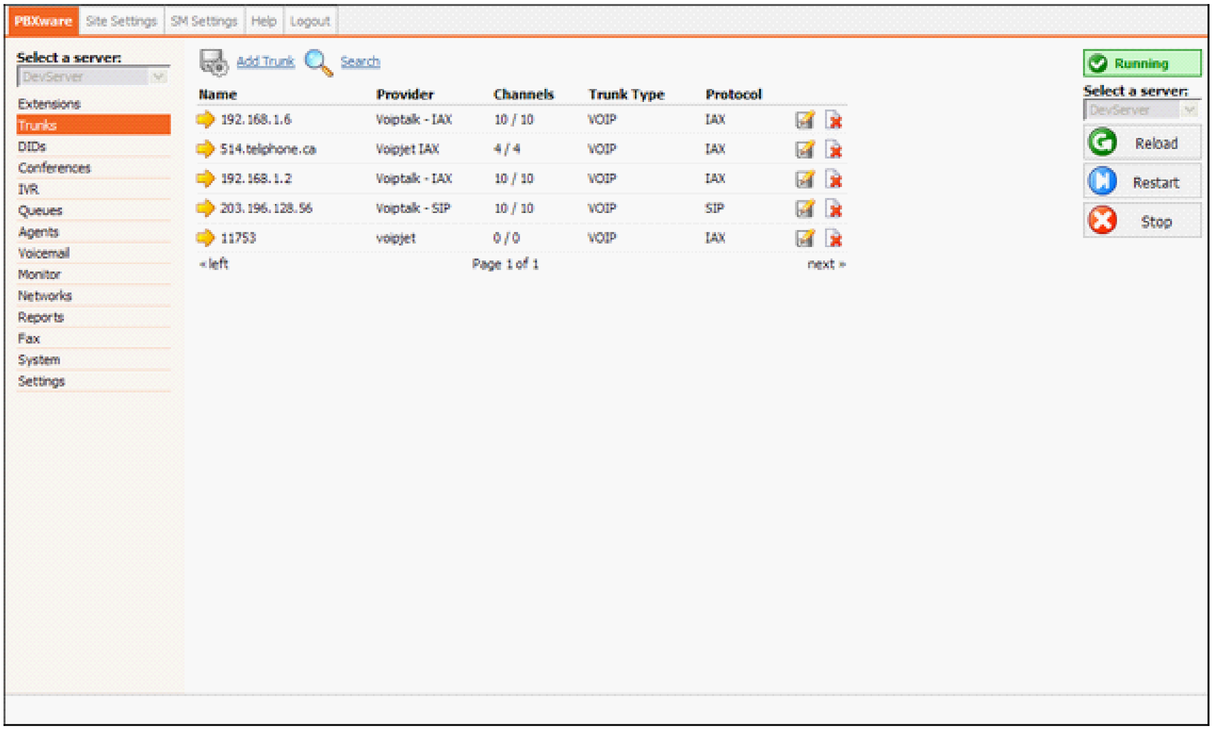
Secret/Password used by the UAD/Phone for the registration with the PBXware

Example: By default this field is automatically populated, but can be changed to any value
Field Type: [a-z][0-9]

Part V

5 Trunks

Trunks are a transmission lines between two systems. This transmission is done using a wide range of PSTN and VOIP technologies. This screen lists all system Trunks with the following details



Name:
Trunk name
Example: TrunkName/192.168.1.6
Field Type: Display

Provider:
Provider template name
Example: SIP TRUNK
Field Type: Display

Channels:
Number of incoming/outgoing channels used by the trunk
Example: 4/4
Field Type: Display

Trunk Type:
Type of a trunk
Example: PSTN/VOIP
Field Type: Display

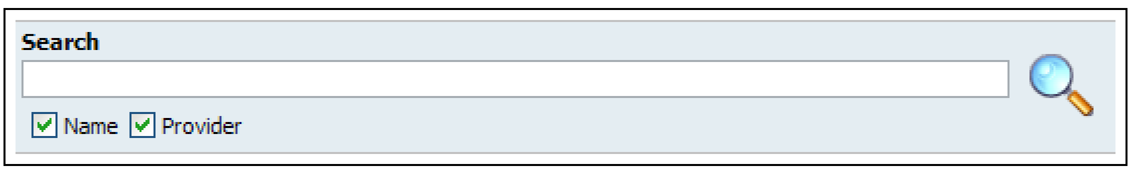
Protocol:
Protocol used by the trunk
Example: SIP/IAX
Field Type: Display

Edits the trunk configuration
Example: Click to edit trunk configuration
Field Type: Button

Deletes a trunk from the system
Example: Click to delete a trunk from the system
Field Type: Button

5.1 Search

Search bar filters trunks by name and provider



Search:
Search phrase
Example: Provide a search phrase here and hit enter to filter the records
Field Type: [a-z][0-9]

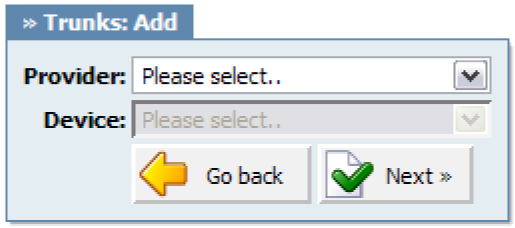
Name:
Should search filter be applied to trunk names
Example: Check the box to search trunk names
Field Type: Check box

Provider:
Should search filter be applied to provider names
Example: Check the box to search provider names
Field Type: Check box

5.2 Add/Edit Trunk

When adding new trunk, first step requires 'Provider' and 'Device' selection.

NOTE: Although new trunk can be created without it, it is preferred that 'Provider' template is created first under 'Settings: Providers'

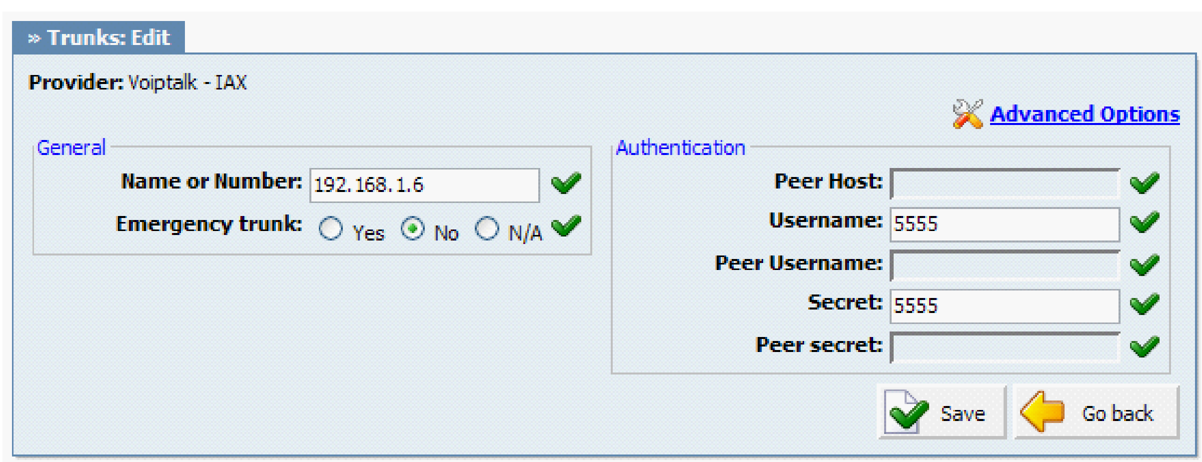


Provider:
Select a service provider template
Example: BT
Field Type: Select box

Device:
If the providers service requires a device in order to provide the service, this field will become visible.
Example: None, T100
Field Type: Select box

5.2.1 VoIP

Second step of the trunk installation and trunk edit command, opens the following options



Name or Number:

Some providers require this field to be equal to DID number (e.g.55510205); but if connecting two systems, IP address may be used as well.

Example: 2554433, myvoiceboxlink

Field Type: [a-z] [0-9]

Emergency trunk:

Should emergency services (Police, Ambulance etc) be dialed through this trunk

Example: Dialling 911 will pass the call through this trunk
Field Type: Option buttons

Peer Host:

IP of a peer host system sends the calls to

Example: 192.168.1.1
Field Type: IP Address

Username:

Username for authenticating with the service provider

Example: 2554433
Field Type: [0-9]

Peer Username:

Peer username for authenticating with the service provider

Example: 2554433
Field Type: [a-z] [0-9]

Secret:

Secret/Password used for authenticating with the service provider

Example: 123456
Field Type: [a-z] [0-9]

Peer secret:

Peer secret/password used for authenticating with the service provider

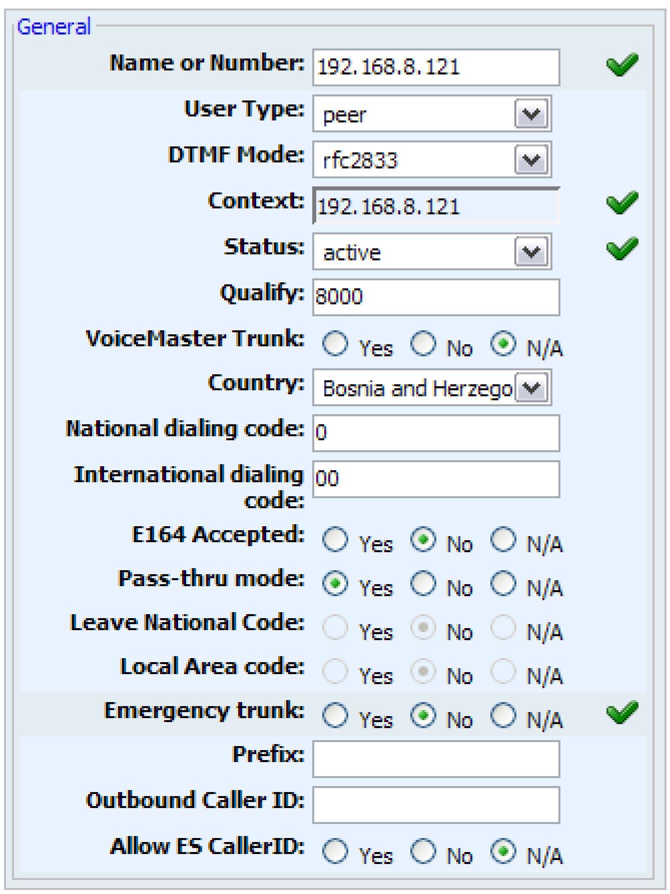
Example: 123456
Field Type: [a-z] [0-9]

5.2.1.1 Advanced Options

In this chapter we will cover:

5.2.1.1.1 General

These options are used frequently and required for normal trunk operation. Some of these fields are pre-configured with the default values. It is not recommended to change them.



User Type:

User's relationship to the system

Example:
- **user** - Trunk accepts incoming calls only
- **peer** - Trunk makes outgoing calls only
- **friend** - Trunk does both incoming and outgoing calls

Field Type: Select box

DTMF Mode (Dual Tone Multi-Frequency):

Trunk DTMF mode. A specific frequency (consisting of two separate tones) to each key so that it can easily be identified by a microprocessor

Example:
- **inband** - inband audio(requires 64 kbit codec - alaw, ulaw)
- **rfc2833** - default
- **info** - SIP INFO messages

Field Type: Select box

Context:

Contexts define a scope within the PBXware. Trunk context cannot be modified and is same as the trunk name or number

Example: 2554433
Field Type: [a-z][0-9]

Status:

Set trunk status on the system

Example: Rather then deleting the trunk you can disable it on system level by selecting 'Not Activated' here
Field Type: Select box

Qualify:

Timing interval in milliseconds at which a 'ping' is sent to a host in order to find out its status

Example: Set this field to 2000 for example. If more time then provided here is needed to reach the host, host is considered offline
Field Type: [0-9]

VoiceMaster Trunk:

Set whether this trunk leads to VoiceMaster gateway

Example: If you have a VoiceMaster gateway and are creating this trunk to connect it with this system, select 'Yes' here
Field Type: Option buttons

Country:

Country where service provider resides

Example: Select USA for example if provider is from United States
Field Type: Select box

National Dialing Code:

National dialing code used at the trunk destination

Example: For USA **1**, United Kingdom, Germany **0**
Field Type: [0-9]

International Dialing Code:

International dialing code used at the trunk destination

Example: For USA **011**, United Kingdom, Germany **00**
Field Type: [0-9]

E164 Accepted:

Does the trunk support dialing destinations in E164 format

Example: Enabling this option will reformat any dialled number into following form COUNTRY_CODE + AREA_CODE + DIALED_NUMBER. For example, if you dial 55510205, system will dial 121255510205
Field Type: Option buttons

Pass-thru mode:

Pass the digits dialed without any conversion (E164, National, Area code). NOTE: When active, 'Leave National Code and 'Local Area Code' will be disabled

Example: If this option is disabled PBXware will convert all dialed numbers to E164 format (COUNTRY_CODE + AREA_CODE + DIALED_NUMBER) and then make a call to converted number. If this option is enabled, PBXware will call directly DIALED_NUMBER without making any number conversions

Field Type: Option buttons

Leave National Code:

In some countries, national code is stripped automatically. If set to 'Yes', national code will not be stripped from the dialed number. NOTE: Before settings this option to 'Yes', go to 'Settings: Servers' and enable this options as well.

Example: John dials 121255510205. With this option enabled
Field Type: [0-9]

Local Area Code:

Add local area code to dialed number, if required by service provider. (By default, local area code is stripped when dialing)

Example: User dials 55510205, local area code is 212. If call goes through this trunk PBXware will dial 21210205
Field Type: [0-9]

Prefix:

Value added to all dialed numbers going over the trunk

Example: Prefix **5**, Dialed number **123**, System dials **5123**
Field Type: [0-9]

Outbound Caller ID:

If Caller ID is not set by UAD, value provided here will be used instead for all outgoing calls

Example: 55599999
Field Type: [0-9]

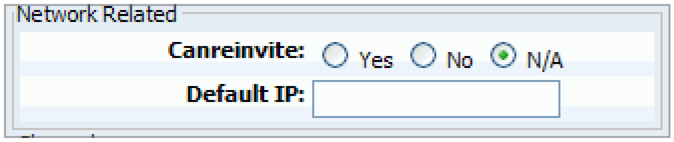
Allow ES Caller ID:

Should ES (Enhanced Services) caller id be allowed over this trunk

Example: Any extension can set custom caller id for each system trunk. With this option enabled, that caller id will be used instead of the trunk outbound caller id
Field Type: [0-9]

5.2.1.1.2 Network Related

These options set important network related values regarding NAT



Canreinvite:

Should you allow RTP voice traffic to bypass PBXware

Example: Some devices do not support this especially if one of them is behind a NAT
Field Type: Options buttons

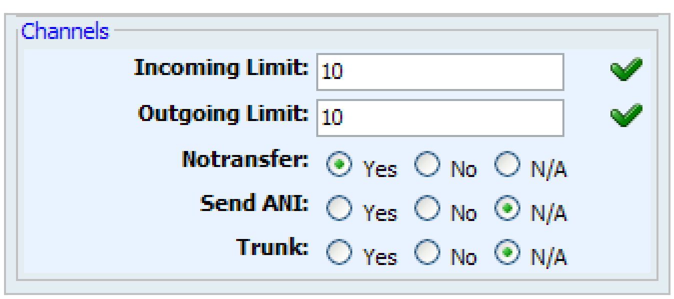
NOTE: All enhanced services for the extension have to be disabled

Default IP:

IP address to be used until registration

Example: 192.168.1.1
Field Type: IP Address

5.2.1.1.3 Channels



Incoming Limit:

Number of simultaneous incoming calls trunk can handle

Example: 4 equals to four simultaneous incoming calls. Any additional calls will get the busy sound
Field Type: [0-9]

Outgoing Limit:

Number of simultaneous outgoing calls trunk can handle

Example: 4 equals to four simultaneous outgoing calls. Any additional calls attempting to use this trunk will be rejected or will be redirected to other trunks depending on what is set in the system/extensions
Field Type: [0-9]

Notransfer:

Disable native IAX transfer

Example:
Field Type: Option buttons

Send ANI:

Should ANI ("super" Caller ID) be sent over this trunk

Example: Set 'Yes' to enable
Field Type: Option buttons

Trunk:

Use IAX2 trunking with this host

Example: Set 'Yes' to enable
Field Type: Option buttons

5.2.1.1.4 Authentication

Host:

IP address of a host trunk is connecting to

Example: Enter host IP, 192.168.1.1 for example or set 'dynamic' if host is behind dynamic IP address
Field Type: [0-9][a-z]

Register:

Method for registering to remote server

Example: Providers may require different way of registration to their server. You may choose between 'registration not required', 'register with phone number' and 'register with username'
Field Type: Select box

Register suffix:

Service provider may request different registration methods for their services. Select the proper method, as required by the provider

Example: 1234567
Field Type: [0-9]

From Domain:

From domain data required by some providers

Example: If your provider requires this information, provide the exact value here
Field Type: [a-z][0-9]

Auth Method:

Authentication method required by provider

Example: md5
Field Type: [a-z] [0-9]

RSA key:

RSA authentication key

Example: If Auth Method is set to RSA, then provide the RSA key here
Field Type: [a-z][0-9]

Encryption:

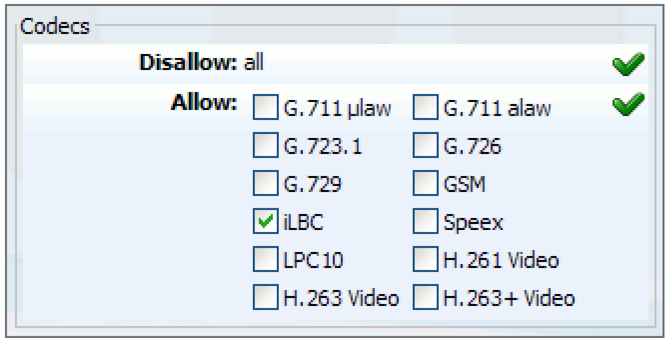
Should encryption be used when authenticating with the peer

Example:
Field Type: [a-z][0-9]

5.2.1.1.5 Codecs

Codecs are used to convert analog to digital voice signals and vice versa. These options set preferred codecs used by the extension.

NOTE: If some of the desired codecs cannot be checked, go to 'Settings: Servers: Edit: Default Codecs' and enable them under the 'Remote' group.



Disallow:

Set the codecs trunk is now allowed to use

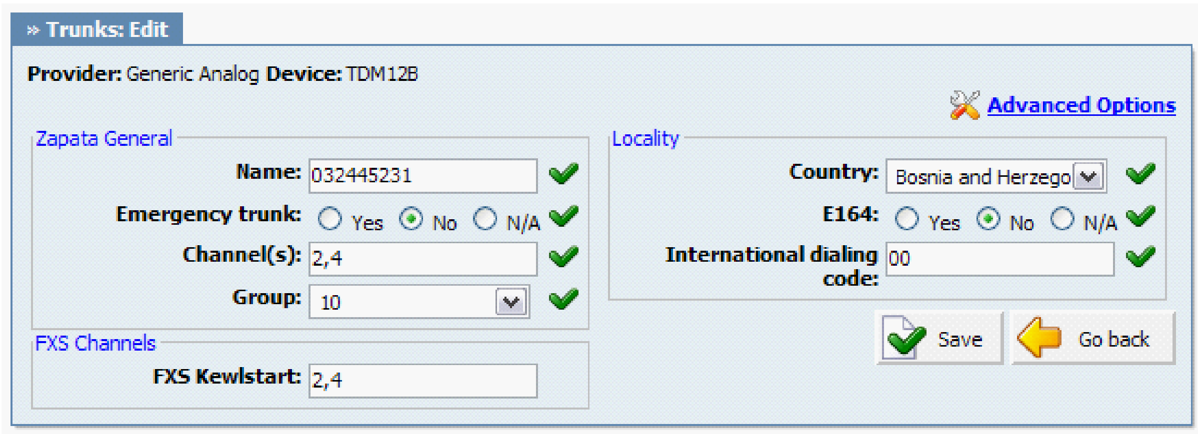
Example: This field is very unique. In order to work properly, this setting is automatically set to 'Disallow All' and it cannot be modified
Field Type: Read only

Allow:
Codecs that are allowed in **'Settings: Server'** will be enabled for selection.
Example: all

- **ITU G.711 ulaw** - 64 Kbps, sample-based, used in US
- **ITU G.711 alaw** - 64 Kbps, sample-based, used in Europe
- **ITU G.723.1** - (5.3/6.3 Kbps, 30ms frame size
- **ITU G.726** - 16/24/32/40 Kbps
- **ITU G.729** - 8 Kbps, 10ms frame size
- **GSM** - 13 Kbps (full rate), 20ms frame size
- **iLBC** - 15Kbps,20ms frame size: 13.3 Kbps, 30ms frame size
- **Speex** - 2.15 to 44.2 Kbps
- **LPC10** - 2.5 Kbps
- **H.261 Video** - Used over ISDN lines with resolution of 352x288
- **H.263 Video** - Low-bit rate encoding solution for video conferencing
- **H.263+ Video** - Extension of H.263 that provides additional features that improve compression over packet switched networks.

Field Type: Check boxes

5.2.2 PSTN



Name:

Trunk name/number

Example: 032445231 for example
Field Type: [a-z][0-9]

Emergency trunk:

Should emergency services (Police, Ambulance etc) be called over this trunk

Example: Select 'Yes' in order to dial emergency services over this trunk
Field Type: Option buttons

Channels

Which card channels are used

Example: If channel 2 and 4 are used on your card, set '2,4' here. If all four channels are used set '1-4' here
Field Type: [0-9], [,-]

Group

Every ZAPTEL trunk has to belong to a group. Selecting any group will enable the trunk

Example: With most of the cards this option is auto detected and set. If that is the case with your card - do not change this field
Field Type: Select box

FXS Kewlstarts:

Signaling protocol for analog circuits that better detects far-end disconnects

Example: Select card channels to be monitored with it. For example '1,4' or '1-4'. These numbers should match the 'Channel(s)' field
Field Type: [0-9], [,-]

Country:

Destination of the trunk connection

Example: If your system is located in USA, select USA here
Field Type: Select box

E164 Accepted:

Does the trunk support dialing destinations in E164 format

Example: Enabling this option will reformat any dialled number into following form COUNTRY_CODE +AREA_CODE+DIALED_NUMBER. For example, if you dial 55510205, system will dial 121255510205
Field Type: Option buttons

International Dialing Code:

International dialing code at the trunk destination

Example: For USA **011**, United Kingdom, Germany **00**
Field Type: [0-9]

5.2.2.1 Advanced

Advanced options section contains all fields which relate to zapata and zaptel configuration files:

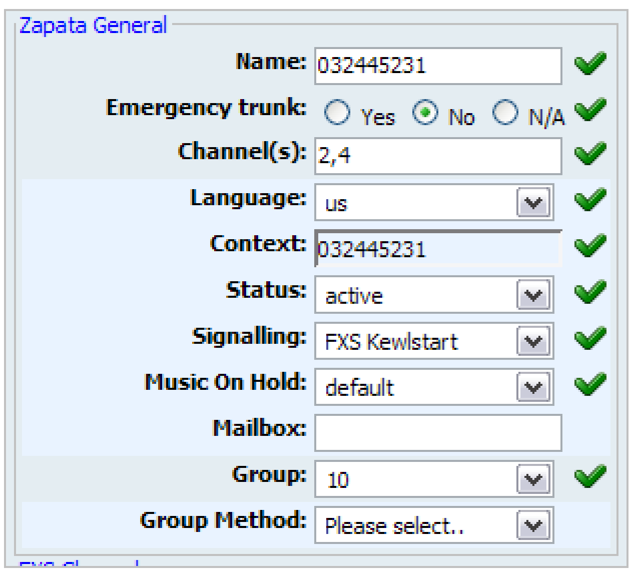
Zapata 128
Configuration of your hardware interface(s)

Zaptel 147
Configuration to use your hardware interface(s)

5.2.2.1.1 Zapata

In this chapter we will cover:

5.2.2.1.1.1 General



Language:

Default language

Example: us
Field Type: Select box

Context:

Contexts define a scope within the PBXware.

Example: default
Field Type: [a-z][0-9]

Status:

Trunk status

Example:
- Active
- Not Activated

Field Type: Select box

Signalling:

Signalling method

Example: default
- FXS Loopstart
- FXS Groundstart
- FXS Kewlstart
- FXO Loopstart
- FXO Groundstart
- FXO Kewlstart
- PRI CPE side
- PRI Network side
- BRI CPE side
- BRI Network side
- BRI CPE PTMP
- BRI Network PTMP

Field Type: Select box

Music On Hold:

Select which class of music to use for music on hold. If not specified then the 'default' will be used

Example: default
Field Type: Select box

Mailbox:

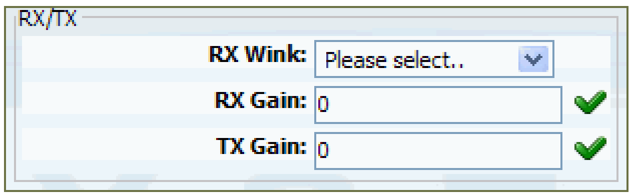
Define a voicemail context

Example: 1234, 1234@context
Field Type: [a-z][0-9]

Group Method:

Example:
Field Type: [a-z][0-9]

5.2.2.1.1.2 RX/TX



RX Wink:

Set timing parameters

Example:
- Pre-wink (50ms)
- Pre-flash (50ms)
- Wink (150ms)
- Receiver flashtime (250ms)
- Receiver wink (300ms)
- Debounce timing (600ms)

Field Type: Select box

RX Gain:

Receive signal decibel

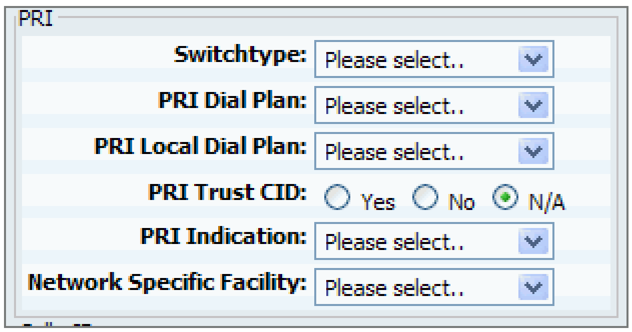
Example: In case that incoming sound is low and you cannot hear other party well, set this option to 2. That should increase incoming sound by 2 decibels
Field Type: [0-9]

TX Gain:

Transmit signal decibel

Example: In case that outgoing sound is low and other party cannot hear you well, set this option to 2. That should increase outgoing sound by 2 decibels
Field Type: [0-9]

5.2.2.1.1.3 PRI



Switchtype:

Set switch type

Example:
- National ISDN 2
- Nortel DMS100
- AT&T 4ESS
- Lucent 5ESS
- EuroISDN
- Old National ISDN 1

Field Type: Select box

PRI Dial Plan:

Set dial plan used by some switches

Example:
- Unknown
- Private ISDN
- Local ISDN
- National ISDN
- International ISDN

Field Type: Select box

PRI Local Dial Plan:

Set numbering dial plan for destinations called locally

Example:
- Unknown
- Private ISDN
- Local ISDN
- National ISDN
- International ISDN

Field Type: Select box

PRI Trust CID:

Trust provided caller id information

Example: Yes, No, N/A
Field Type: Option buttons

PRI Indication:

How to report 'busy' and 'congestion' on a PRI

Example:
- **outofband -** Signal Busy/Congestion out of band with RELEASE/DISCONNECT
- **inband -** Signal Busy/Congestion using in-band tones

Field Type: Select box

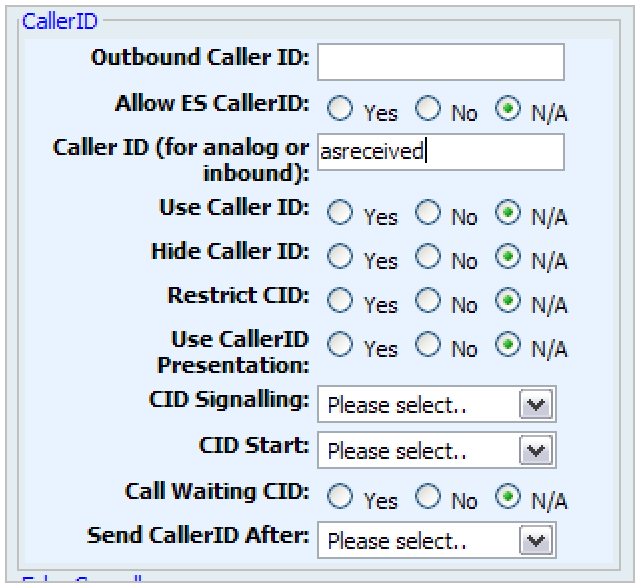
Network Specific Facility:

If required by switch, select network specific facility

Example:
- none
- sdn
- megacom
- accunet

Field Type: Select box

5.2.2.1.1.4 Caller ID



Outbound Caller ID:

Caller ID set for all outbound calls where Caller ID is not set or supported by a device

Example: john@domain.com
Field Type: [0-9]

Allow ES Caller ID:

Should ES (Enhanced Services) Caller ID be allowed over this trunk

Example: Any extension can set custom Caller ID for each system trunk. With this option enabled, that Caller ID will be used instead of the Trunk Outbound Caller ID
Field Type: [0-9]

Caller ID (for analog or inbound):

CallerID can be set to 'asreceived' or a specific number if you want to override it

Example: 'asreceived', 555648788
Field Type: [a-z][0-9]

NOTE: Caller ID can only be transmitted to the public phone network with supported hardware, such as a PRI. It is not possible to set external caller ID on analog lines

Use Caller ID:

Whether or not to use caller id

Example: Yes, No, N/A
Field Type: Options buttons

Hide Caller ID:

Whether or not to hide outgoing caller ID

Example: Yes, No, N/A
Field Type: Options buttons

Restrict CID:

Whether or not to use the caller ID presentation for the outgoing call that the calling switch is sending

Example: Yes, No, N/A
Field Type: Options buttons

Use CallerID Presentation:

Whether or not use the caller ID presentation for the outgoing call that the calling switch is sending

Example: -
Field Type: Options buttons

CID Signalling:

Set the type of caller ID signalling

Example:
- **bell** - US
- **v23** - UK
- **dtmf** - Denmark, Sweden and Netherlands

Field Type: Select box

CID Start:

What signals the start of caller ID

Example:
- ring = a ring signals the start
- polarity = polarity reversal signals the start

Field Type: Select box

Call Waiting CID:

Whether or not to enable call waiting on FXO lines

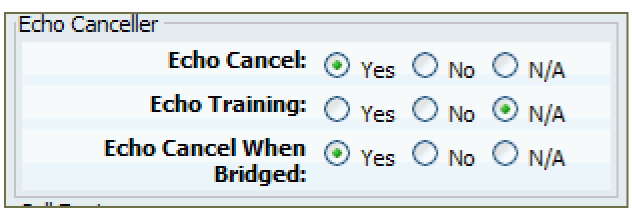
Example: Yes, No, N/A
Field Type: Options buttons

Send CallerID After:

Some countries, like UK, have different ring tones (ring-ring), which means the caller id needs to be set later on, and not just after the first ring, as per the default.

Example: Yes
Field Type: Select box

5.2.2.1.1.5 Echo Canceller



Echo Cancel:

Enable echo cancellation

Example: Yes, No, N/A
Field Type: Option buttons

Echo Training:

Mute the channel briefly, for 400ms, at the beginning of conversation, cancelling the echo. (Use this only if 'Echo Cancel' doesn't work as expected)

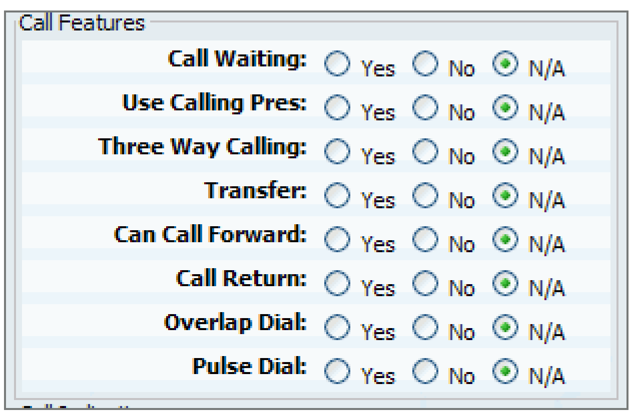
Example: Yes, No, N/A
Field Type: Option buttons

Echo Cancel When Bridged:

Enable echo cancellation when bridged. Generally not necessary, and in fact undesirable, to echo cancel when the circuit path is entirely TDM

Example: Yes, No, N/A
Field Type: Option buttons

5.2.2.1.1.6 Call Features



Call Waiting:

Whether or not to enable call waiting on FXO lines

Example: Yes, No, N/A
Field Type: Option buttons

Use Calling Pres:

Whether or not use the caller ID presentation for the outgoing call that the calling switch is sending

Example: Yes, No, N/A
Field Type: Option buttons

Three Way Calling:

Support three-way calling. If enabled, call can be put on hold and one is able to make another call

Example: Yes, No, N/A
Field Type: Option buttons

Transfer:

Support call transfer and also enables call parking (overrides the 'canpark' parameter). Requires 'Three Way Calling' = 'Yes'.

Example: Yes, No, N/A
Field Type: Option buttons

Can Call Forward:

Support call forwarding

Example: Yes, No, N/A
Field Type: Option buttons

Call Return:

Whether or not to support Call Return **'*69'.** Dials last caller extension number

Example: Yes, No, N/A
Field Type: Option buttons

Overlap Dial:

Enable overlap dialing mode (sends overlap digits)

Example: Yes, No, N/A

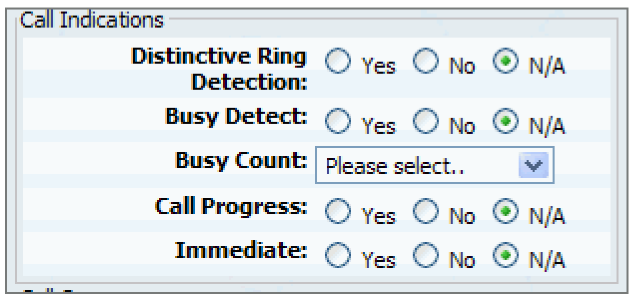
Field Type: Option buttons

Pulse Dial:

Use pulse dial instead of DTMF. Used by FXO (FXS signalling) devices

Example: Yes, No, N/A
Field Type: Option buttons

5.2.2.1.1.7 Call Indications



Distinctive Ring Detection:

Whether or not to do distinctive ring detection on FXO lines

Example: Yes, No, N/A
Field Type: Options buttons

Busy Detect:

Enable listening for the beep-beep busy pattern

Example: Yes, No, N/A
Field Type: Options buttons

Busy Count:

How many busy tones to wait before hanging up. Bigger settings lower probability of random hangups. 'Busy Detect' has to be enabled

Example:

- 4
- 6
- 8

Field Type: Select box

Call Progress:

Easily detect false hangups

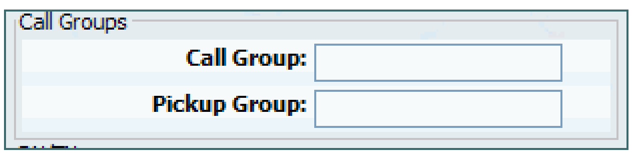
Example: Yes, No, N/A
Field Type: Options buttons

Immediate:

Should channel be answered immediately or the simple switch should provide dialtone, read digits, etc

Example: Yes, No, N/A
Field Type: Options buttons

5.2.2.1.1.8 Call Groups



Call Group:

Which group is allowed to pickup incoming calls by dialing *8. The default value is empty.

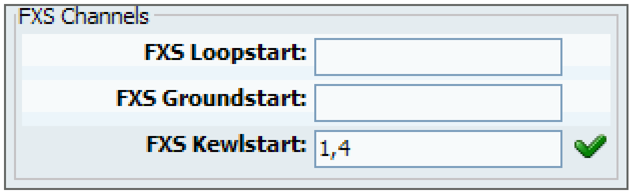
Example: 1, 1-4
Field Type: [0-9]

Pickup Group:

Which groups are allowed to pickup calls by dialing *8. The default value is empty.

Example: 1, 1-4
Field Type: [0-9]

5.2.2.1.1.9 FXS Channels



FXS Loopstart:

Signals the far end that it wants the dial tone by shorting the leads

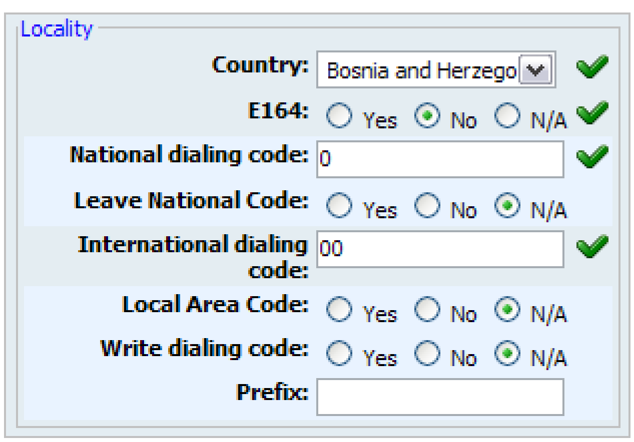
Example: default
Field Type: [a-z][0-9]

FXS Groundstart:

Signals the far end that it wants the dial tone by grounding one of the leads

Example: default
Field Type: [a-z][0-9]

5.2.2.1.1.10 Locality



Country:

Destination of the trunk connection

Example: USA
Field Type: Select box

E164 Accepted:

Does the trunk support dialing destinations in E164 format

Example: Enabling this option will reformat any dialled number into following form COUNTRY_CODE +AREA_CODE+DIALED_NUMBER. For example, if you dial 55510205, system will dial 121255510205
Field Type: Option buttons

National Dialing Code:

National dialing code at the trunk destination

Example: For USA **1**, United Kingdom, Germany **0**
Field Type: [0-9]

Leave National Code:

In some countries, national code is stripped automatically. If set to 'Yes', national code will not be stripped from the dialed number. NOTE: Before settings this option to 'Yes', go to 'Settings: Servers' and enable this options as well.

Example: John dials 121255510205. With this option enabled
Field Type: [0-9]

International Dialing Code:

International dialing code at the trunk destination

Example: For USA **011**, United Kingdom, Germany **00**
Field Type: [0-9]

Local Area Code:

Add local area code to dialed number, if required by service provider. (By default, local area code is stripped when dialing)

Example: User dials 55510205, local area code is 212. If call goes through this trunk PBXware will dial 21210205
Field Type: [0-9]

Write dialing code:

Should National and International prefix be written into configuration files

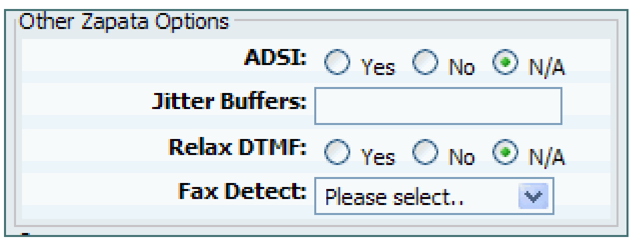
Example: Enable this option if required by provider
Field Type: [0-9]

Prefix:

Value added to all dialed numbers going over the trunk

Example: Prefix **5**, Dialed number **123**, System dials **5123**
Field Type: [0-9]

5.2.2.1.1.11 Other Zapata Options



ADSI (Analog Display Services Interface):

Enable remotely controlling of screen phone with softkeys. (Only if you have ADSI compatible CPE equipment)

Example: Yes, No, N/A
Field Type: Option buttons

Jitter Buffers:

Configure jitter buffers. Each one is 20ms long

Example: 4
Field Type: [0-9]

Relax DTMF:

If you are having trouble with DTMF detection, you can relax the DTMF detection parameters

Example: Yes, No, N/A
Field Type: Option buttons

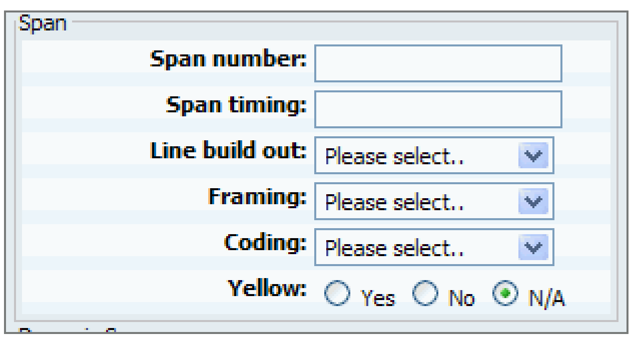
Fax Detect:

Enable fax detection

Example:
- both
- incoming
- outgoing
- no

Field Type: Select box

5.2.2.1.1.12 Span



Span number:

Number of the span

Example: 1
Field Type: [0-9]

Span timing:

How to synchronize the timing devices

Example:
- 0 - do not use this span as sync source
- 1 - use as primary sync source
- 2 - set as secondary and so forth

Field Type: [a-z]

Line build out:

Example:
- 0 db (CSU) / 0-133 feet (DSX-1)
- 133-266 feet (DSX-1)
- 266-399 feet (DSX-1)
- 399-533 feet (DSX-1)
- 533-655 feet (DSX-1)
- -7.5db (CSU)
- -15db (CSU)
- -22.5db (CSU)

Field Type: Select box

Framing:

How to communicate with the hardware at the other end of the line

Example:
- For T1: Framing is one of d4 or esf.
- For E1: Framing in one of cas or ccs.

Field Type: Select box

Coding:

How to encode the communication with the other end of line hardware.

Example:
- For T1: coding is one of ami or b8zs
- For E1: coding is one of ami or hdb3 (E1 may also need crc)

Field Type: Select box

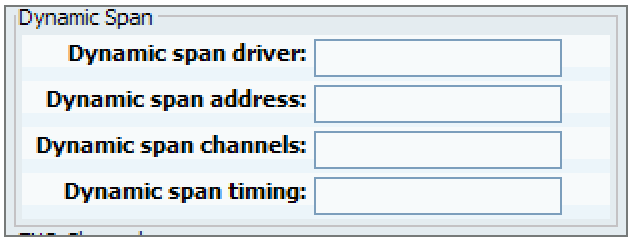
Yellow:

Whether yellow alarm is transmitted when no channels are open.

Example:
- Yes
- No
- N/A

Field Type: Option buttons

5.2.2.1.1.13 Dynamic Span



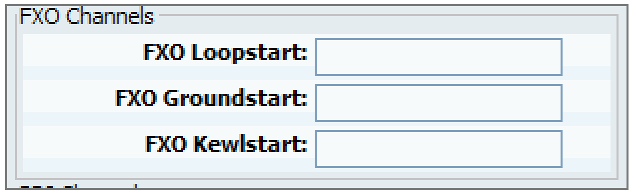
Dynamic span driver:
The name of the driver (e.g. eth)

Dynamic span address:
Driver specific address (like a MAC for eth).

Dynamic span channels:
Number of channels.

Dynamic span timing:
Sets timing priority, like for a normal span. Use "0" in order not to use this as a timing source, or prioritize them as primary, secondary, etc.

5.2.2.1.1.14 FXO Channels



FXO Loopstart:
Channel(s) are signalled using FXO Loopstart protocol

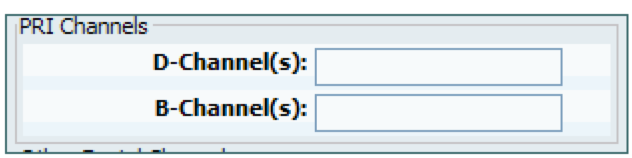
FXO Groundstart:
Channel(s) are signalled using FXO Groundstart protocol

FXO Kewlstart:
Channel(s) are signalled using FXO Kewlstart protocol

Values for above fields are set as follows:

- 1 - for one card
- 1-2 - for two cards
- 1-3 - for three cards etc or
- 2-3 (If your card has modules in this order FXS, FXO, FXO, FXS)

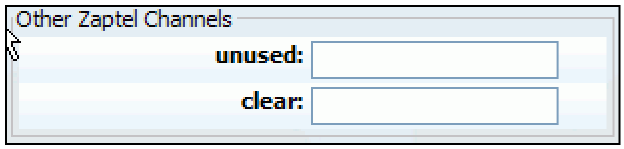
5.2.2.1.1.15 PRI Channels



D-Channel(s):
For example, every ISDN BRI card has 1 D- (control) channel
Example: 1
Field Type: [0-9]

B-Channels(s):
For example, every ISDN BRI card has 2 B- (data) channels
Example: 2
Field Type: [0-9]

5.2.2.1.1.16 Other Zaptel Channels



Unused:
Example:
Field Type: [0-9]

Clear:
Example:
Field Type: [0-9]

5.2.2.1.2 Zaptel

The zaptel.conf file is where the TDM-specific interface parameters required for ZAPTEL card(s) can be configured.

Span number
Span number set which span this trunk is to use.

Span timing
Span timing sets where is zaptel going to obtain its timing from. Choices are: 0 (Telco), 1 (System) , 2 (?)

Line Build Out
Line build out sets the distance between service hardware equipment.

Framing
Framing used on this trunk.

Coding
Coding used on this trunk

Yellow
Yellow field can be set to "Yes" or "No"

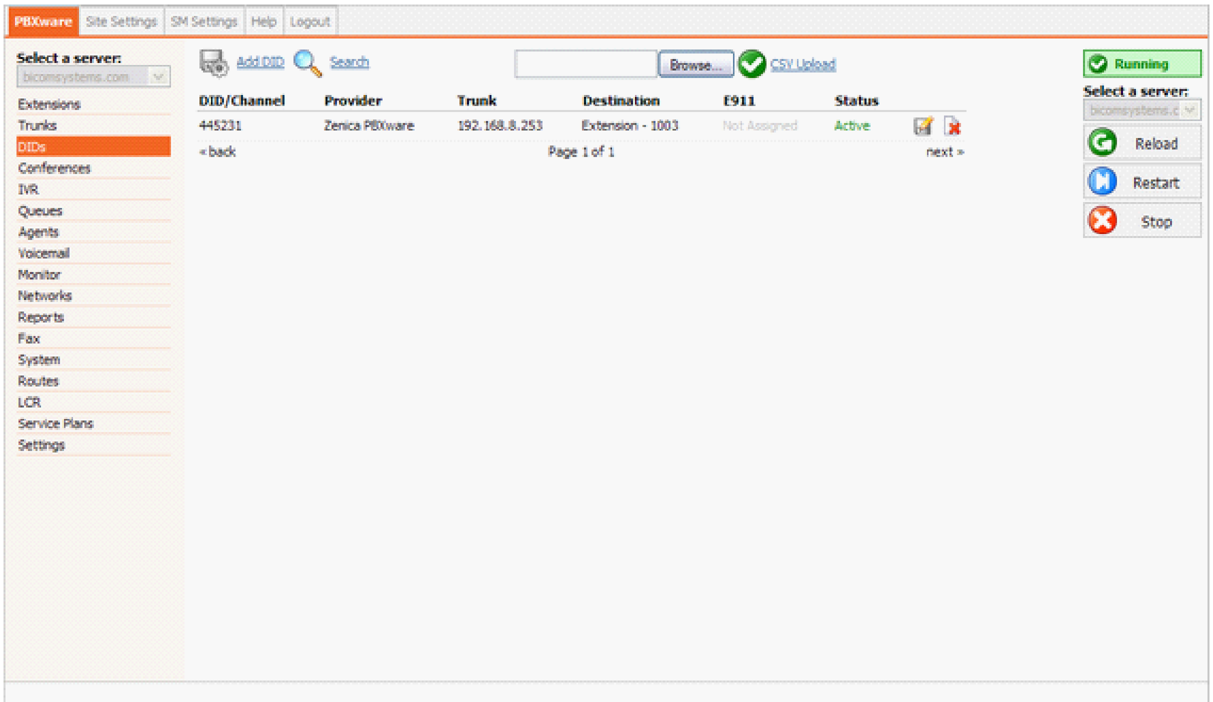
Signalling, D and B Channels
fields: "e&m, fxsls, fxsgs, fxsks, fxols, fxogs, fxoks, sf, unused, clear, indclear, rawhdlc, dchan, bchan, fcshdlc, nethdlc, dacs"
are used to set signalling and operation mode for one or range of channels

Zones
Choose zone your interface card will be loaded with indication tones of.
defaultzone=us
loadzone=$VALUE (For Example: **us**)
Available zones are: us, au, fr, nl, uk, fi, es, jp, no, at, nz, it, us-old, gr, tw, cl

Part VI

6 DIDs

DIDs are used to point all incoming calls (that come over trunks) to specific system destinations. This screen lists all system DIDs with the following details



DID/Channel:
DID number or a PSTN channel slot number
Example: 1000/1
Field Type: Display

Provider:
Provider name
Example: SIP TRUNK
Field Type: Display

Trunk:
Trunk used by a DID
Example: 192.168.1.6/TrunkName
Field Type: Display

Destination:
Trunk destination and destination network number
Example: Network User - 5000
Field Type: Display

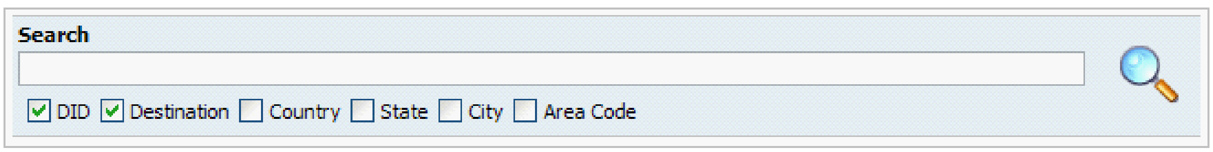
Status:
DID status
Example: Active/Inactive
Field Type: Display

Edits the DID configuration
Example: Click to edit the DID configuration
Field Type: Button

Deletes a DID from the system
Example: Click to delete a DID from the system
Field Type: Button

6.1 Search

By selecting 'Search' Command, search menu will be displayed. Searches can be done by DID value, Trunk name, Provider name and Destination value



Search:
Search phrase
Example: Provide a search phrase here and hit enter to filter the records
Field Type: [a-z][0-9]

DID:
Should search filter be applied to DID values
Example: Check the box to search DID values
Field Type: Check box

Destination:
Should search filter be applied to DID destinations
Example: Check the box to search DID destinations
Field Type: Check box

Country:
Should search filter be applied to country field
Example: Check the box to search countries
Field Type: Check box

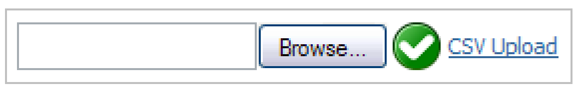
State:
Should search filter be applied to state field
Example: Check the box to search states
Field Type: Check box

City:
Should search filter be applied to city field
Example: Check the box to search cities
Field Type: Check box

Area Code:
Should search filter be applied to area code field
Example: Check the box to search area codes
Field Type: Check box

6.2 CSV Upload

To create multiple DIDs on the fly all you have to do is to upload a .csv file with DID details. Click on 'Browse' button, select a .csv file from your computer, click 'Open' then 'CSV Upload' button.



NOTE: CSV file must be in the following format:

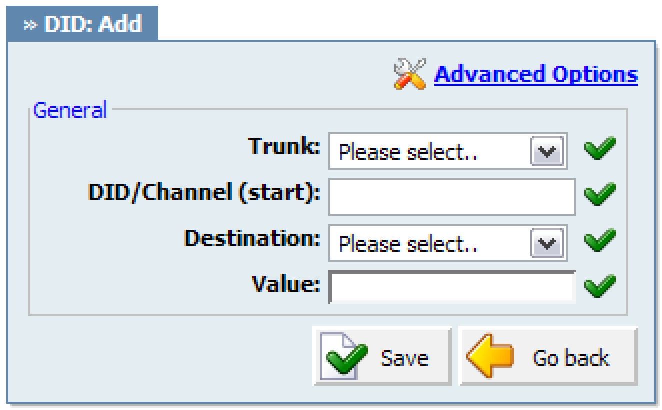
'number1,number2,type,extension,callerid' or '1080,1084,SUB,1002,7777'

DID Type can be set to:

- SUB - Subscriber/Extensions
- MUL - Multi user
- IVR - Interactive Voice Response
- QUE - Queue
- VML - Voicemail
- RMT - Remote Destination
- CNF - Conference
- FAX

6.3 Add/Edit DID

A click on 'Add/Edit' will open standard DID options



Trunk:

Select the trunk DID will pickup the calls from

Example: If you select '2554433' for example, DID will wait for any incoming calls over that trunk and then will pass the call further based on the settings below
Field Type: Select box

DID/Channel (start):

Provide a DID number here (e.g. 55510205)

Example: If selected 'Trunk' is PSTN or VOIP, set the line number here
Field Type: [0-9]

Destination:

PBXware destination DID will transfer all calls to (Extensions, IVRs, Queues, Voicemails, Remote Access, Conferences and even to Fax to Email service)

Example:
Field Type: Select box

Value:

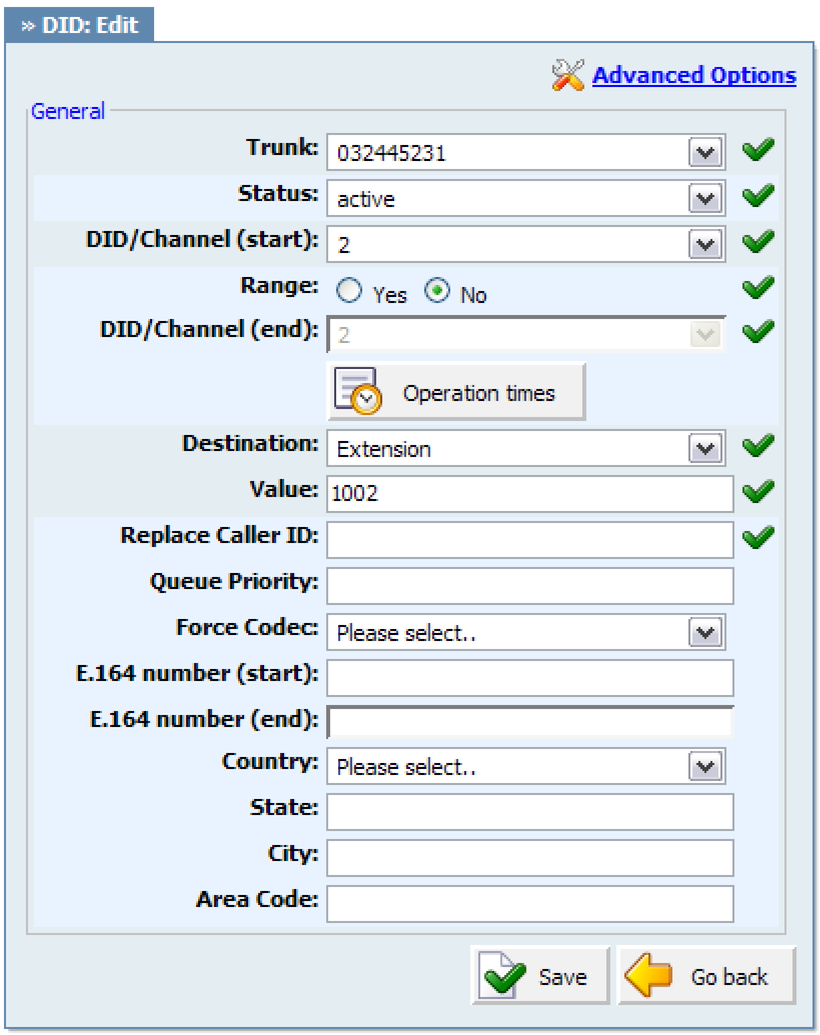
Destination extension all DID calls will be transferred to.

Example: If 'Destination'='Extension', set the extension number here(e.g. 1002). In case 'Destination'='IVR', set the IVR extension number here
Field Type: [0-9]

NOTE:
If using 'Fax to email', set this field to email address (email@domain.com) or extension number (1002). If extension number is used fax will go to email address associated to extension.
In order to send fax remotely set this field to 'remote:fax'

6.4 Advanced Options

A click on 'Advanced Options' button will open more detailed DID options



Status:

Set the DID status on the network.

Example: Rather then deleting the DID you can deactivate it by selecting 'Not Activated' and restore it back with 'Active'
Field Type: Select box

Range:

Some providers offer a range of numbers over a single trunk. Set whether this DID should be used to transfer a range of numbers to some PBXware destination.

Example: John has bought 10 DID numbers from a provider (55510205 - 55510215) and wants all calls coming from these to be transferred to lobby queue. He needs to set this options to 'Yes'. Set 55510205 to (start), 55510215 to (end) fields. Set 'Destination'='Queues' and 'Value'=Queue number(e.g. 1000).
Field Type: Options buttons

DID/Channel (end):

Some providers offer a range of numbers over a single trunk. Set whether this DID should be used to transfer a range of numbers to some PBXware destination.

Example: John has bought 10 DID numbers from a provider (55510205 - 55510215) and wants all calls coming from these to be transferred to lobby queue. He needs to set 'Range'='Yes'. Set 55510205 to (start) and 55510215 to this field. Set 'Destination'='Queues' and 'Value'=Queue number(e.g. 1000).
Field Type: Options buttons

Operation Times:

Set the DID operation time

Example: For more explanation click here
Field Type: Option buttons

Replace Caller ID:

Replaces the caller id with the custom data provided here. This is used when you want all incoming DID calls to have this value displayed as a caller id information.

Along with the custom data, you can use the '%PRODUCT%' variable, which displays the calling party phone number.

NOTE: Please make sure you enter this information as it is written down, otherwise, it will not work properly.

Example: Providing a 'USDID' here, will display 'USDID' on your phone display, for all calls coming through this DID.

Providing 'USDID %PRODUCT%', will display 'USDID 55510205' on your phone display, where 55510205 is calling party phone number.
Field Type: [a-z][0-9] [%CALLERID%]

Queue Priority:

Set Queue priority

Example: If this DID redirects all calls to queue, set '1' here to give all calls over this trunk the highest queue priority
Field Type: [a-z][0-9]

Force Codec:

Force a codec to all calls going over this trunk

Example: Select G.711 ulaw from the list to force all calls going over this DID to this codec
Field Type: [a-z][0-9]

E.164 number (start):

A DID number in E.164 format 'INTERNATIONAL PREFIX + AREA CODE + PHONE NUMBER' (1 212 555 9876).

If provided here, this number will be used by 'PBXware: Networks' and will be dialed over the Internet rather then PSTN trunk. If 'Range' field is set to 'Yes' provide the DID/Channel (start) number in E.164 format here

Example: If your DID number = 5559876, and you live in NewYork/USA, your E.164 number is 12125559876
Field Type: [0-9]

E.164 number (end):

A DID number in E.164 format 'INTERNATIONAL PREFIX + AREA CODE + PHONE NUMBER' (1 212 555 9876).

If 'Range' field is set to 'Yes' provide the DID/Channel (end) number in E.164 format here

Example: If your DID number = 5559876, and you live in NewYork/USA, your E.164 number is 12125559876
Field Type: [0-9]

Country:

Select a country this DID number belongs to

Example: If DID number is in USA format (e.g. 1212****) select USA here
Field Type: [a-z][0-9]

State:

State DID number belongs to

Example: If DID number is in USA format (e.g. 1212****), 212 is in New York so type NY here
Field Type: [a-z][0-9]

City:

City DID number belongs to

Example: If DID number is in USA format (e.g. 1212****), 212 is in New York so type New York here
Field Type: [a-z][0-9]

Area Code:

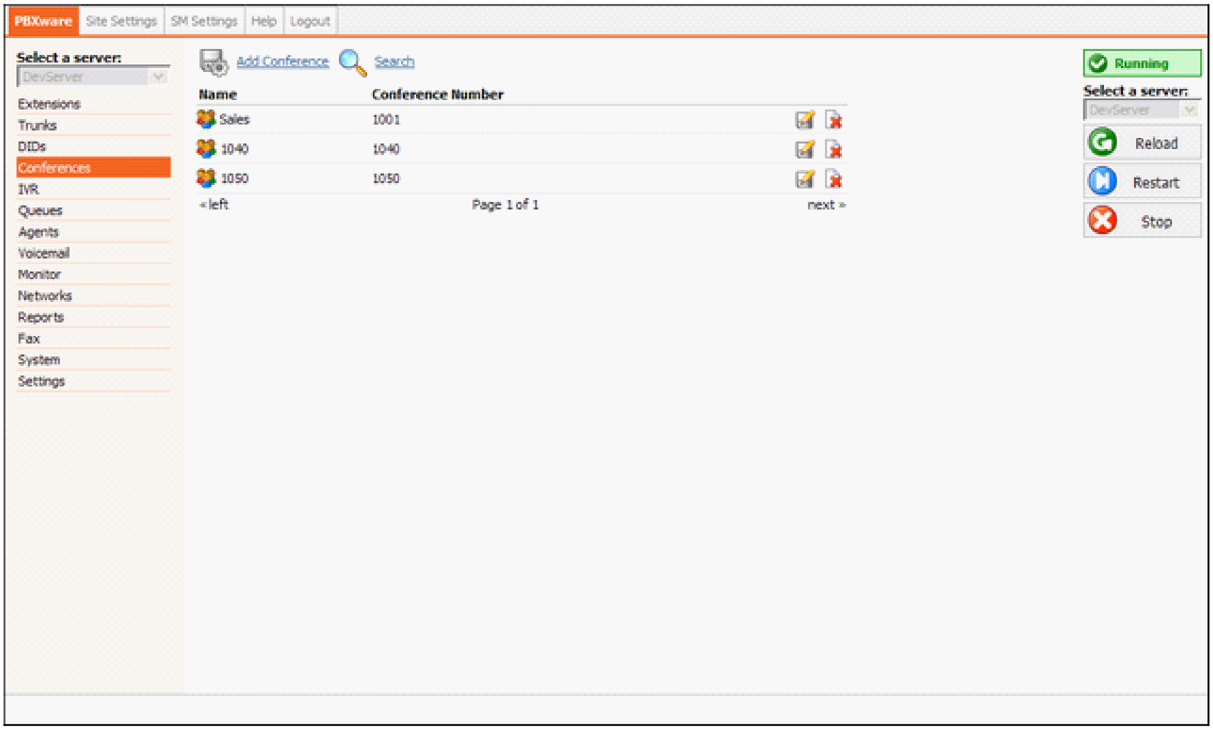
Area code DID number belongs to

Example: If DID number is in USA format (e.g. 1212****), type 212 for that is the area code for New York
Field Type: [0-9]

Part VII

7 Conferences

Conferences allow two or more participants to communicate with each other at the same time using voice, video or both. This screen lists all system conferences with the following details



Name:
Conference name
Example: Sales, Development
Field Type: Display

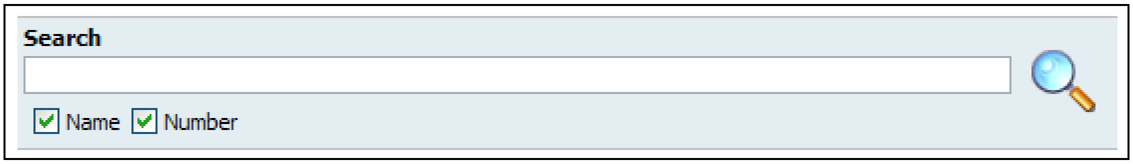
Conference Number:
Conference system or network number
Example: 2255
Field Type: Display


Edits the Conferences configuration
Example: Click to edit Conference configuration
Field Type: Button


Deletes a Conference from the system
Example: Click to delete a Conference from the system
Field Type: Button

7.1 Search

By selecting 'Search' Command, search menu will be displayed. Searches can be done by Name and Number



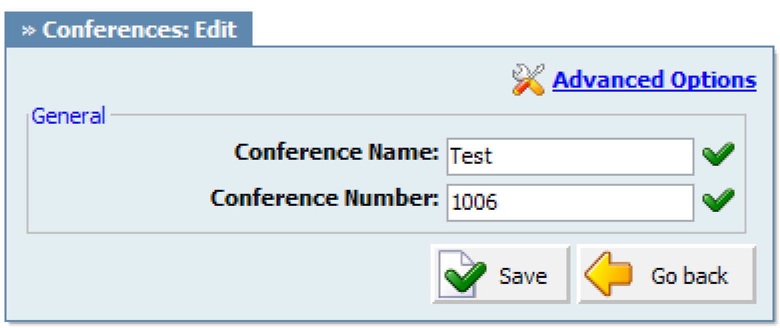
Search:
Search phrase
Example: Provide a search phrase here and hit enter to filter the records
Field Type: [a-z][0-9]

Name:
Should search filter be applied to conference names
Example: Check the box to search conferences names
Field Type: Check box

Number:
Should search filter be applied to conference numbers
Example: Check the box to search conference numbers
Field Type: Check box

7.2 Add/Edit Conference

A clicking on 'Add/Edit' button will open standard conferences options



Conference Name:

Unique Conference identifier/name

Example: This name will be displayed once 'Conferences' menu is selected
Field Type: [a-z] [0-9]

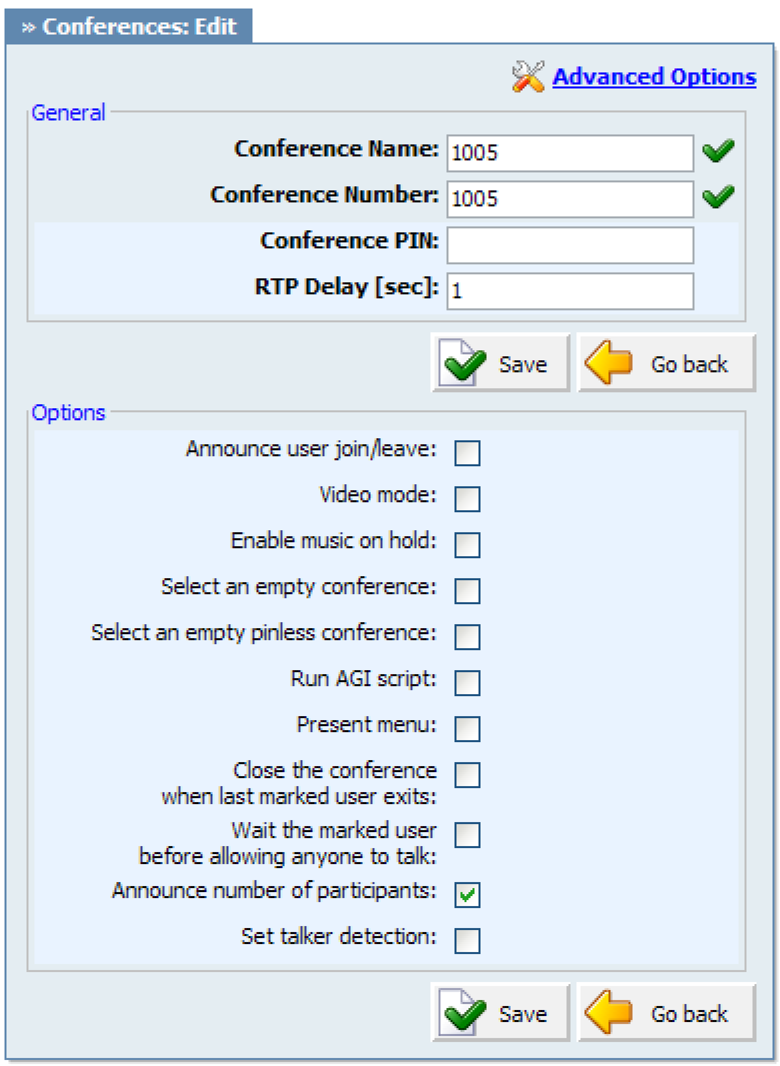
Conference Number:

Unique Conference PBXware number

Example: This number is to be dialed in order to access the conference
Field Type: [0-9]

7.3 Advanced Options

A click on 'Advanced Options' button will open more detailed conferences options



Conference PIN:

Conference PIN number. If set, this PIN will have to be provided by all callers before entering the Conference.

Example: John dials the sales conference (1001), but this conference has this field set to 9576. John is asked to provide the conference PIN. If valid PIN is provided, John will enter the conference, or his call will be rejected after 3 invalid PIN entries.
Field Type: [0-9]

RTP Delay [sec]:

Delay time in seconds inserted before the conference operator answers. This delay solves the 'half-played' greeting file problem. Keep this value set between 1-3 for optimal performance.

NOTE: The 'half-played' greeting file problem usually exists on VOIP trunks.

Example: John enters the sales conference (1001) and hears '...e currently the only user in this conference' message. This message is only partial and can confuse conference members. Set this field to 1 (or higher, depending on how much is not heard) to play the message in full length.
Field Type: [0-9]

Announce user join/leave:

If this option is checked/enabled, all new conference members will be asked to say their name and press the '#' key before they enter the conference. This name will be recorded and played to other conference members when caller joins/leaves the conference.

Example: John dials 1001 sales conference and is asked to say his name and to press the '#' key. Of course, he says 'John' and presses the '#' key. At the same time, all conference members will hear 'John has entered the conference'. When John leaves, all conference members will hear 'John has left the conference' message.
Field Type: Select box

Video mode:

Enable video conferencing

Example: This option will enable video conferencing if UAD/Phone support it.
Field Type: Select box

Enable music on hold:

Enable MOH(Music On Hold) if there is a single member in a conference.

Example: John enters the conference and he is the only one there. Don't let John feel alone. Enable this feature to play MOH class music files until someone else joins the conference :)
Field Type: Select box

Select an empty conference:

If this option is enabled, all callers trying to enter this conference will be redirected to a new empty conference, if such empty conference exists. If empty conference does not exist, the caller will join the conference he dialed.

Example: John dials 1001 sales conference but this conference has this option enabled. If PBXware has another conference with no members in it, John will be transferred to it.
If sales conference that John dialed is the only PBXware conference, or if there are other conferences but with members in them, John will be allowed to enter the dialed sales conference.
Field Type: Select box

Select an empty pinless conference:

If this option is enabled, all callers trying to enter this conference will be redirected to a new empty PIN-less conference, if such empty conference exists. If empty PIN-less conference does not exist, the caller will join the conference he dialed.

Example: John dials 1001 sales conference but this conference has this option enabled. If PBXware has another PIN-less conference with no members in it, John will be transferred to it.
If sales conference that John dialed is the only PBXware conference, or if there are other PIN-less conferences but with members in them, John will be allowed to enter the dialed sales conference.
Field Type: Select box

Run AGI script:

Runs the custom AGI script ('conf-background.agi') once the caller enters the conference.

NOTE:
- The conference must have at least one zaptel channel to run the script
- This feature is not supported by default in PBXware. Please contact the customer support.

Example: John dials 1001 sales conference but before he enters the conference, a custom AGI script is executed in the background.
Field Type: Select box

Present menu:

Returns the Conference options once * is dialed while in the conference

Example: John enters the sales conference and dials '*'. Conference options are played back (e.g. 'Please press 1 to mute/un-mute yourself').

Field Type: Select box

Close the conference when last marked user exits:

Closes the conference once the last marked user exits, no matter how many participants are still active in the conference conversation; their calls get immediately dropped.

NOTE: Marked user mode is set under 'Extensions: Edit: Conferences: Edit'.

Example: John (marked user) enters the sales conference. This conference has this option enabled and there are 3 more members participating in the conference conversation. As soon as John leaves the conference, all other conference members will have their calls dropped and will no longer be able to talk to each other.
Field Type: Select box

Wait the marked user before allowing anyone to talk:

Disables the conference conversation until the marked user enters the conference.

NOTE: Marked user mode is set under 'Extensions: Edit: Conferences: Edit'.

Example: This option is enabled for sales conference. There are 3 members waiting for John (marked user) to enter the conference. These 3 members will hear nothing, and will not be able to talk to each other until the John enters the sales conference.
Field Type: Select box

Announce number of participants:

Announces the number of conference participants to a new conference member. There is currently only 1 other participant in the conference.

Example: Sales conference has this option enabled and is currently empty. John enters the conference and hears 'You are currently the only person in this conference' payed back to him.
Field Type: Select box

Set talker detection:

Enable the talker detection which is sent to manager interface and conferences List

Example: -
Field Type: Select box

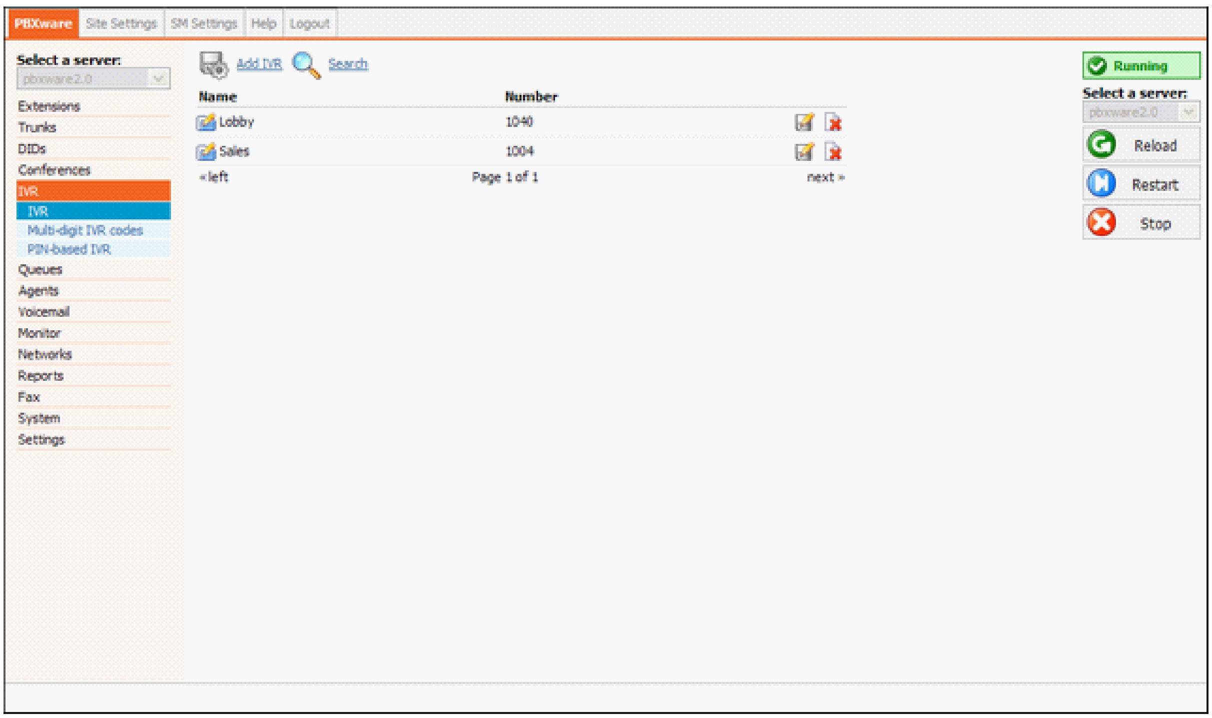
Part VIII

8 IVR

IVRs are automated answering machines which guide callers to his destination by providing a number of choices and waiting for a caller response (dialled digit).

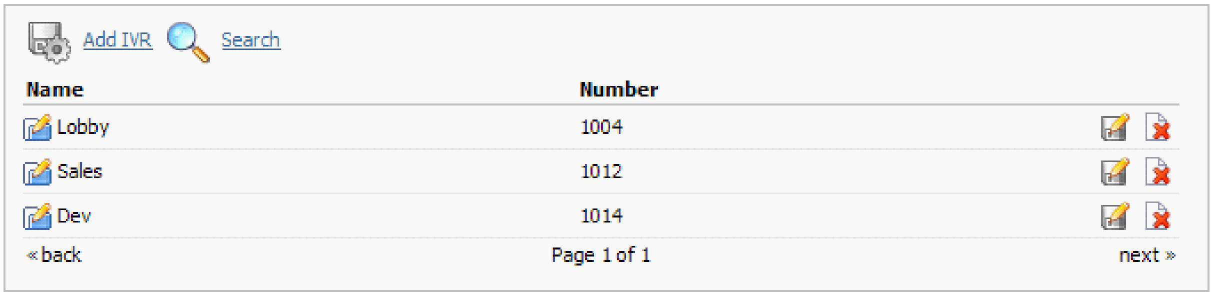
In this chapter we will cover:

- IVR 166
- Multi-digit IVR codes 176
- PIN-based IVR 178



8.1 IVR

This screen lists all system standard DIDs with the following details:



Name:
IVR name
Example: Welcome
Field Type: Display

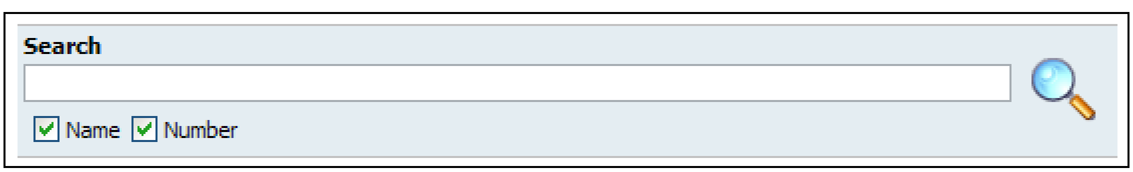
Number:
IVR network number
Example: 1010
Field Type: Display


Edits the IVR configuration
Example: Click to edit IVR configuration
Field Type: Button


Deletes an IVR from the system
Example: Click to delete an IVR from the system
Field Type: Button

8.1.1 Search

By selecting 'Search' Command, search menu will be displayed. Searches can be done by Name and Number



Search:

Search phrase

Example: Provide a search phrase here and hit enter to filter the records
Field Type: [a-z][0-9]

Name:

Should search filter be applied to IVR names

Example: Check the box to search IVR names
Field Type: Check box

Number:

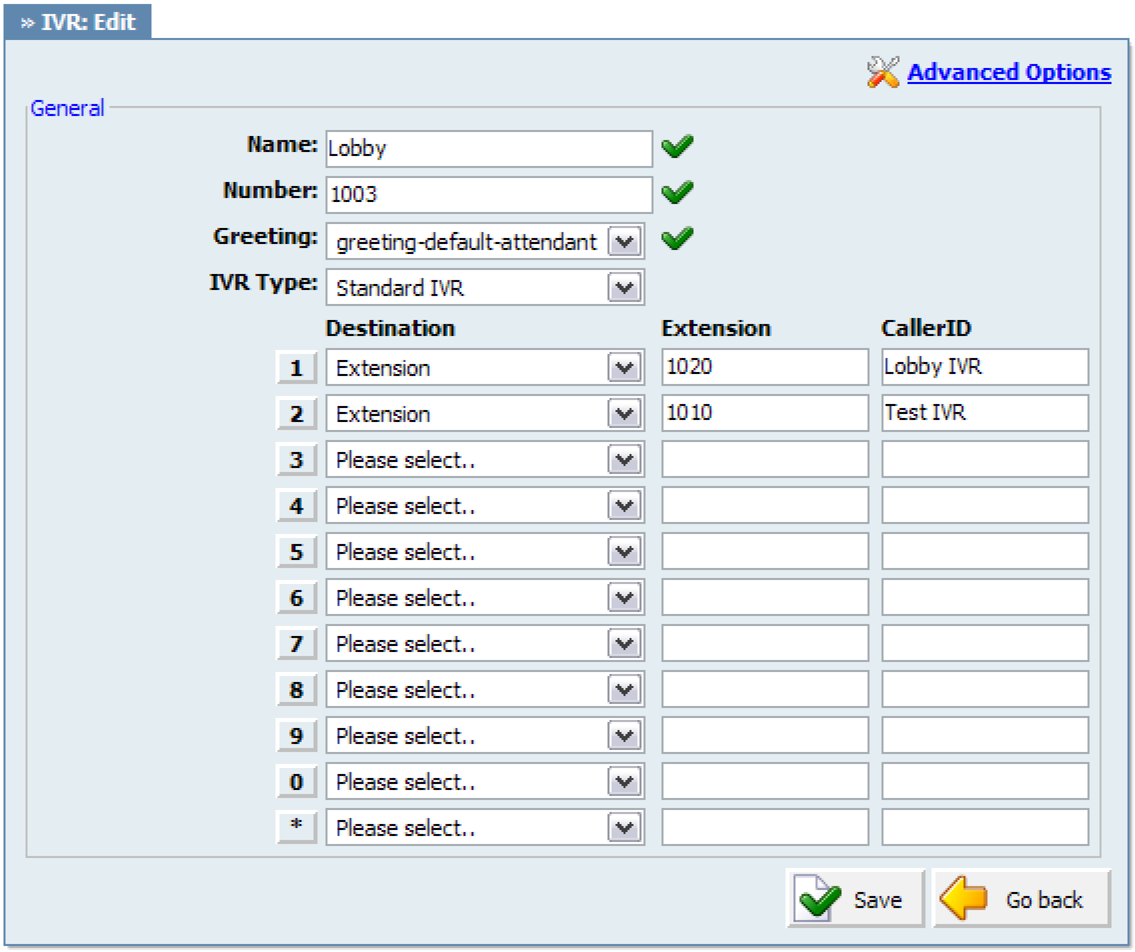
Should search filter be applied to IVR numbers

Example: Check the box to search IVR numbers
Field Type: Check box

8.1.2 Add/Edit IVR

A click on 'Add/Edit' button will open standard IVR options

NOTE: Make sure to create a greeting sound before adding a new IVR. You may create one by dialing '*301' from your UAD/Phone or by uploading a custom sound file from your computer through 'System: Sounds'



Name:

Unique IVR identifier/name

Example: This name will be displayed once IVRs are accessed
Field Type: [a-z] [0-9]

Number:

Unique network IVR number

Example: This number is to be dialed in order to access the IVR
Field Type: [0-9]

Greeting:

Greeting sound file

Example: Once user enters the IVR, a greeting with instructions is played(e.g. 'Welcome. For Sales Press 1...'). Select the greeting file played by this IVR here. NOTE: Greeting file name must start wit 'greeting-***'. To record a custom greeting message dial '*301' from your extension. Newly recorded greeting file will have the current date stamp in the title(e.g. 'greeting-Apr-14-2006-16-32').
Field Type: Select box

IVR Type:

Set the proper IVR type

Example: PBXware works with two type of IVRs: Single and Multi digit ones. Single digit IVR is used for small range of options(0-9). Multi digit IVR support between(10-9999999999) and is shared with all Multi Digit IVRs.
Field Type: Select box

Destination:

Set the proper destination for each digit pressed

Example: Once a greeting message(e.g. 'Press 1 for Sales') is played to user, provide the valid destination where the call is to go to once 1 is pressed. If John from sales department is to be dialed, select 'Extension' in this field. If you wish to provide additional options to caller, you can point him to another IVR with its set of options by selecting 'IVR' here.
Field Type: Select box

Extension:

This field further describes the 'Destination' field. In case 'Remote Access' or 'Queue' are selected under 'Destination', a predefined options will be available for selection under this option.

Example: In the example above we have set the PBXware destination. In this option we set which destination part is to be dialed exactly. If 'Destination'='Extension', provide the extension number here. If 'Destination'='IVR' provide the IVR number here etc...
Field Type: [0-9]

Caller ID:

Overrides the incoming Caller ID with custom information

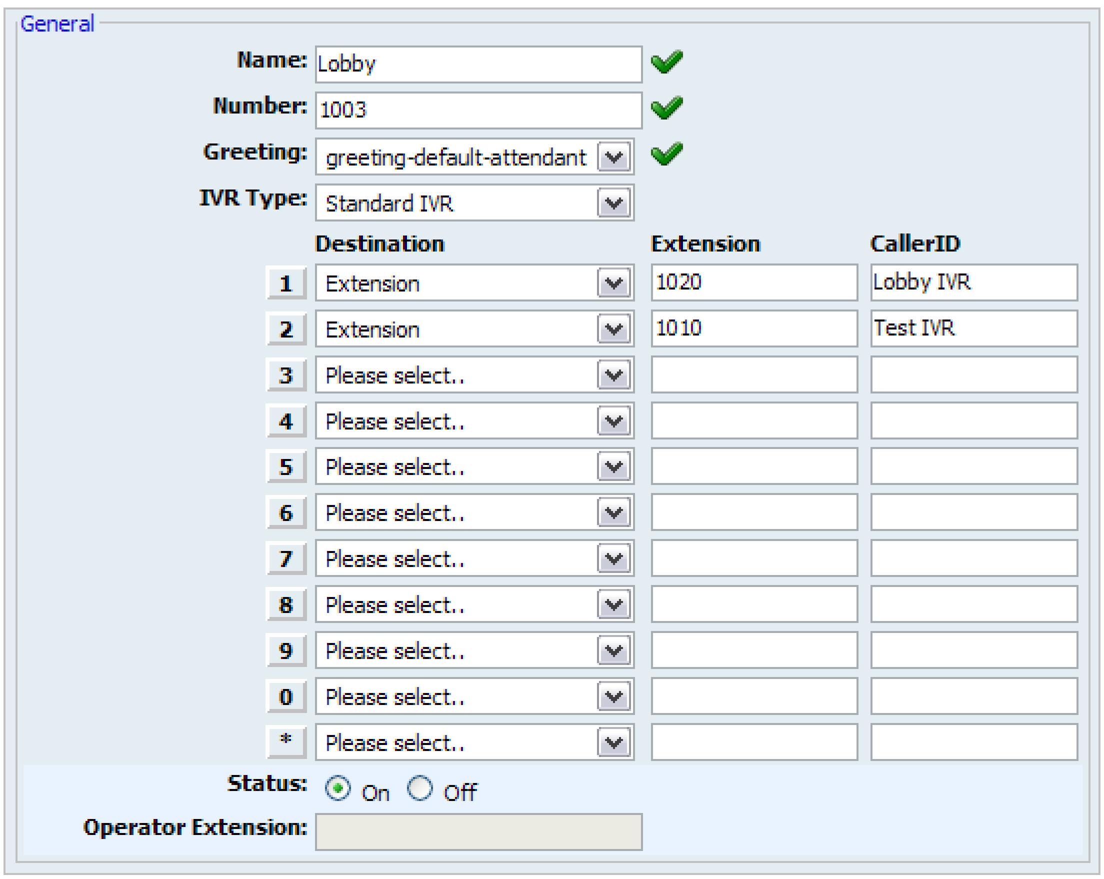
Example: Sometimes, it is useful to know from which IVR the call is coming from. By settings 'Lobby IVR' here, all calls coming through this IVR will display 'Lobby IVR' on phone display. To show the actual phone number along with our data use '%CALLERID%' with our text(e.g. 'Lobby IVR %CALLERID%'). This will display 'Lobby IVR 55528790' on our phone display, where 55528790 is the phone number of the person calling us.
Field Type: [a-z][0-9], [%CALLERID%]

8.1.3 Advanced Options

A click on 'Advanced Options' button will open more detailed IVR options

- General 171
- General Settings 171
- Greeting Options 173
- Operations Times
- Extensions Ringing 174
- Local Dialling 174
- Permissions 175

8.1.3.1 General



Status:

Rather then deleting the IVR, set its status to 'Off'. This will make the IVR inactive and all calls will be transferred to 'Operator Extension'.

Example: Lobby IVR has this option set to 'Off'. John dials this IVR number (e.g. 1003) but instead of IVR instructions, his call will be transferred to 'Operator Extension'
Field Type: Option buttons

Operator extension:

Provide the operator extension to which all calls will be redirected to if 'IVR Status' = 'Off'.

Example: Lobby IVR has the 'Status' set to 'Off'. John dials this IVR but instead of IVR instructions, his call will be transferred to the extension number provided here
Field Type: [0-9]

8.1.3.2 General Settings

Response Timeout:

Time period in seconds during which an IVR option must be dialed by user

Example: John enters the Sales IVR and hears the instructions. If this field is set to '4', John will have 4 seconds to dial an IVR option
Field Type: [0-9]

RTP Delay [sec]:

Delay time in seconds inserted before the IVR greeting message is played. This solves the 'half-played' file problem. Keep this value between 1-3

Example: User A enters the IVR and hears a message '..me. For sales press 1' and wonders what was that.. what did the voice say!? Set this field to 1 so that 1 second pause is added before the message is played. Now, when user A enters the IVR he will hear 'Welcome. For Sales press 1'.
Field Type: [0-9]

Digit Timeout:

Timeout in seconds during which new digit must be dialed. This option is used with Multi-digits IVR.

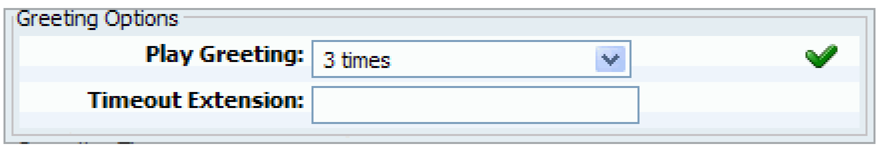
Example: John has entered the IVR and wants to dial option 25. If 1 is provided in this field, John will have 1 second to dial number 2, and additional 1 second to dial number 5. If the time exceeds, and John hits 5 too late, IVR will assume that John has dialed option 2 instead of 25.
Field Type: [0-9]

Rings to Answer:

Number of rings played to caller before a call is allowed to enter the IVR

Example: Rather then just 'falling' into IVR, it is recommended to set the number of ring sounds played to caller
Field Type: [0-9]

8.1.3.3 Greeting Options



Play Greeting:

Number of times greeting message is played to caller. If there is no response from the calling party within this time, the call is disconnected.

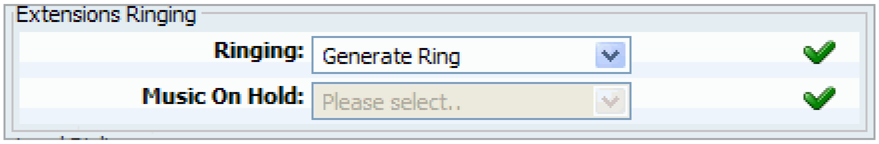
Example: John enters the sales IVR and hears the IVR options. If John does not dial one of the options, IVR options sound file will be played again, a number of times set in this field, before the call gets transferred to 'Timeout Extension'.
Field Type: [0-9]

Timeout Extension:

Extension number to which the IVR call will be transferred to if there is no response from the user during the 'Play Greeting' time period.

Example: John enters the sales IVR and hears the IVR options. If John does not dial one of the options, IVR options sound file will be played again, a number of times set in 'Play Greeting' field, before the call gets transferred to extension number provided here.
Field Type: [0-9]

8.1.3.4 Extensions Ringing



Ringing:

Select the ringing type played back to calling party before they enter the IVR

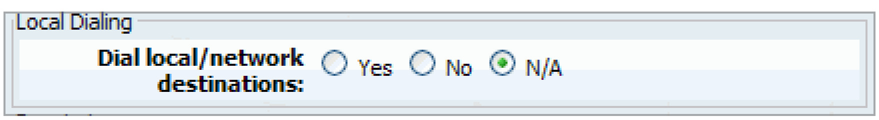
Example: Rather then just falling into the IVR, play the ring sound to user or music files located under the MOH class
Field Type: Select box

Music on Hold:

Select the MOH/Music on Hold class, played to users before they enter the IVR

Example: MOH class usually contains one or more sound files. To see these files go to 'System: MOH'
Field Type: Select box

8.1.3.5 Local Dialing



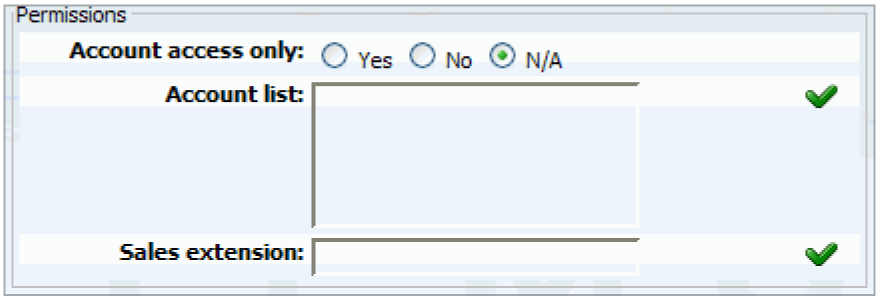
Dial local/network destinations:

IVR option can dial local network or proper/mobile phone numbers. By setting this option to 'Yes' IVR will be allowed to dial local network extensions only

Example: If IVR has this options set to 'Yes', only local network extensions will be accessible from this IVR. No proper/mobile numbers would be dialed.
Field Type: Option buttons

8.1.3.6 Permissions

Permissions is used to allow an organization to restrict who is able to enter an IVR. In particular there are organizations where access to the IVR is only allowed to the callers with a valid account number but it can be used for other similar purposes.



Account Access Only:

Allow only certain PBXware extensions to access the IVR

Example: If this option is set to 'Yes', only extension numbers set under the 'Account list' will be allowed to access this IVR
Field Type: Option buttons

Account List:

Only local extension numbers provided here, separated with a single space, will be allowed to enter this IVR

Example: John dials this IVR, but his extension number is not in the 'Account list'. John is transferred to 'Sales extension'.
Field Type: [0-9]

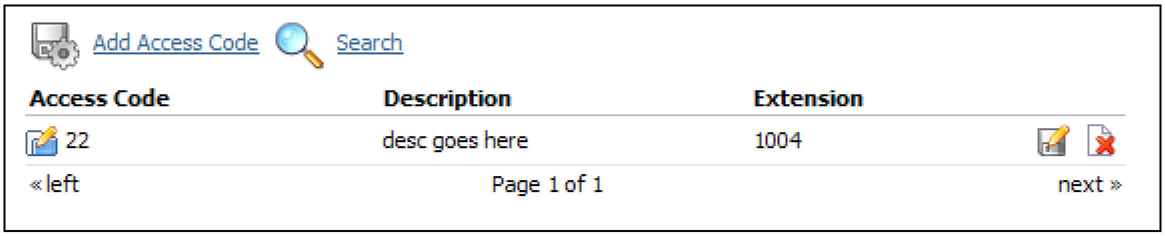
Sales Extension:

If caller extension is not provided in the 'Account list' his call will not enter the IVR, but will be redirected to extension number provided here.

Example: John dials this IVR, but his extension number is not in the 'Account list'. John is transferred to extension number provided here.
Field Type: [0-9]

8.2 Multi-digit IVR codes

The only difference between standard and multi-digit IVRs is that the latter accepts two or more digits as a respons (numbers between 10-100 for example), therefore providing a wider range of options. This screen lists all system Multi-digit DIDs with the following details:



Access Code:
Multi-digit IVR access code
Example: 22
Field Type: Display

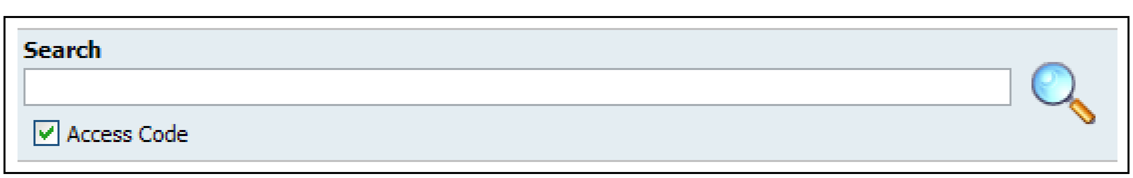
Description:
Access code description
Example: desc goes here
Field Type: Display

Extension:
Extension where call will be transferred to
Example: 1004
Field Type: Display


Edits the IVR configuration
Example: Click to edit IVR configuration
Field Type: Button


Deletes an IVR from the system
Example: Click to delete an IVR from the system
Field Type: Button

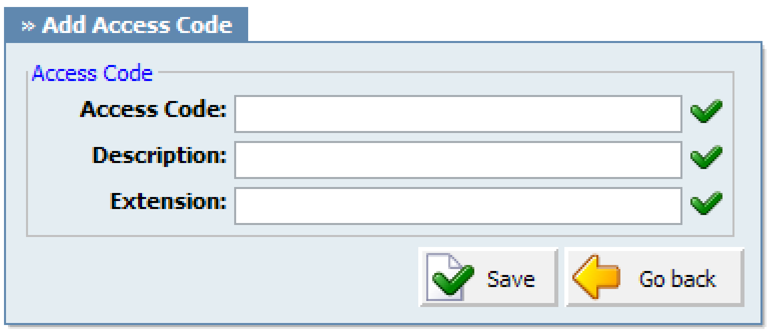
8.2.1 Search



Search:
Search phrase
Example: Provide a search phrase here and hit enter to filter the records
Field Type: [a-z][0-9]

Access Code:
Should search filter be applied to access codes
Example: Check the box to search access codes
Field Type: Check box

8.2.2 Add/Edit Access Code



Access code:

Option dialed from the Multi-digit IVRs that will transfer user to PBXware 'Extension' number.

Example: John enters the Multi-digit IVR and dials this 'Access Code' (e.g. 99). When John dials 99, he will be transferred to PBXware 'Extension' number
Field Type: [0-9]

Description:

A short Description of this access code and its destination. This information is view only by the Administrator

Example: Once the administrator goes to the 'PBXware: Multi-digit IVR codes', a list of access codes, their descriptions and destination extensions will be displayed.
Field Type: [0-9]

Extension:

PBXware extension to which a call will be transferred to once the 'Access code' is dialed from the Multi-digit IVR

Example: John enters the multi-digit IVR and dials 99, which is set to lead to PBXware extension 1000.

When John dials 99, he will be transferred to PBXware 'Extension' provided here
Field Type: [0-9]

8.3 PIN-based IVR

PIN-based IVR allows dialing local/remote destinations by providing a pre-set IVR PIN number

PIN:
IVR PIN number
Example: 55555
Field Type: Display

Description:
IVR PIN short description
Example: Remote Call
Field Type: Display

Expiry date:
PIN expiry date
Example: 31-01-2007
Field Type: Display

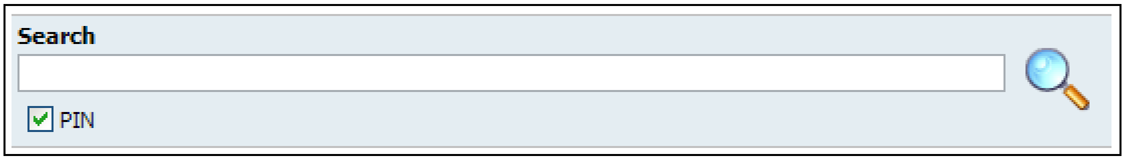
Destination:
Destination dialed once the IVR PIN is provided
Example: 061505139
Field Type: Display


Edits the IVR PIN configuration
Example: Click to edit IVR PIN configuration
Field Type: Button


Deletes IVR PIN settings from the system
Example: Click to delete IVR PIN settings from the system
Field Type: Button

8.3.1 Search

By selecting 'Search' Command, search menu will be displayed. Searches can be done by PIN numbers

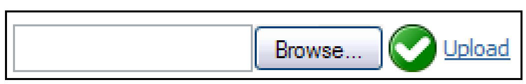


Search:
Search phrase
Example: Provide a search phrase here and hit enter to filter the records
Field Type: [a-z][0-9]

PIN:
Should search filter be applied to PIN numbers
Example: Check the box to search PIN numbers
Field Type: Check box

8.3.2 Upload

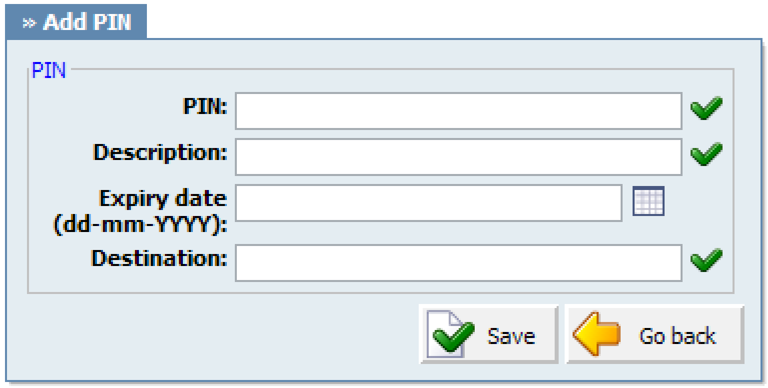
Your may upload a .csv file with multiple PIN and Destination codes in one step. Click the 'Browse' button and select the .csv file on your computer. Then just click the 'Upload' button to add new PIN codes to PBXware.



Sample .csv file:
10002,John Smith Cell,31-01-2007,061555109
20002,George Nimara Home,31-01-2007,032246509

8.3.3 Add/Edit PIN

To manually add IVR PIN code, click the 'Add PIN' button on top. The following options will be displayed.



PIN:

Unique IVR PIN number. This PIN number is provided once requested by IVR. Correctly supplied PIN will make PBXware dial the 'Destination' number.

Example: John dials local IVR (1003) and is asked to provide a PIN. He enters PIN (55555) and PBXware dials the number provided under 'Destination' field.
Field Type: [0-9]

Description:

Short description of IVR PIN. This description is used for describing the Destination number for example

Example: 'John's cell phone', or '21255510205'
Field Type: [a-z] [0-9]

Expiry date:

All PIN codes can be valid until certain date. You are strongly encouraged to set the expiry date by clicking on the 'Calendar' icon, next to the field and selecting a desired date. Date can be provided manually in following form 'dd-mm-yyyy'. Accessing PIN after the Expiry date will be impossible.

Example: If this field is set to expire on 31-01-2007. All IVR calls made with this PIN by this date will be passed through the Destination number. Calls after expire date will not be made.
Field Type: [0-9]

Destination:

Ten digit destination number dialed once the IVR PIN is provided

Example: Provide a 10 digit number here e.g. 1555102057. Once the IVR PIN is provided, a call will be made to it

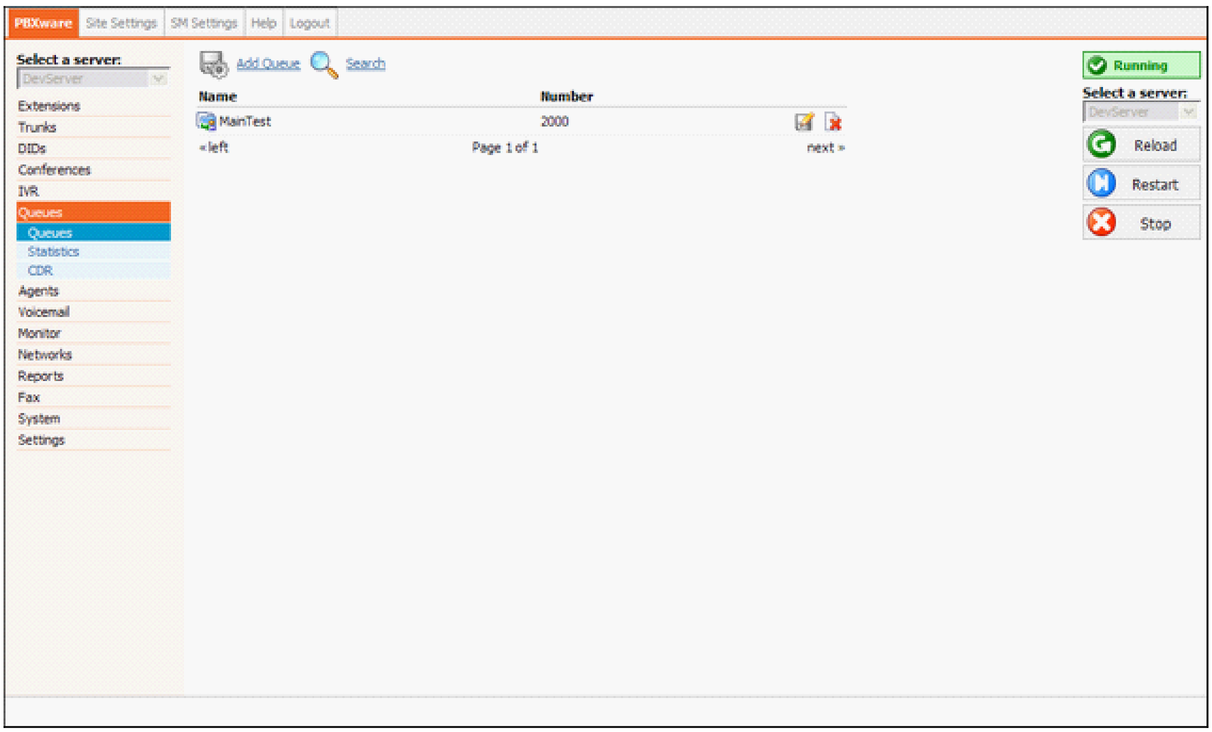
Field Type: [0-9]

Part IX

9 Queues

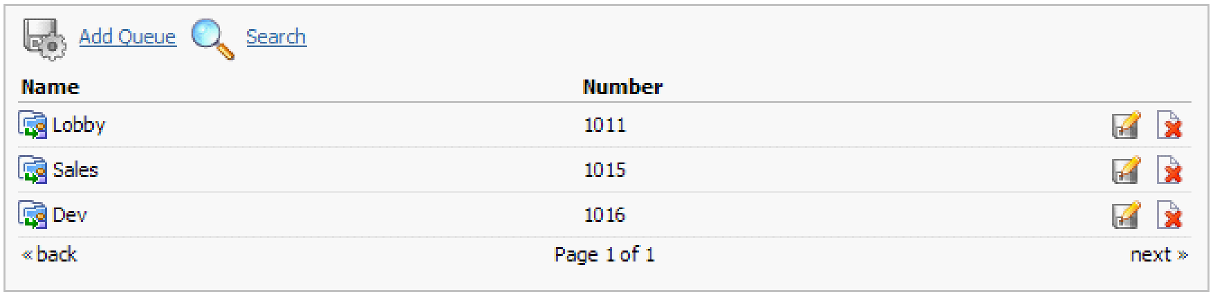
Queue is a place where calls are stack up and where they wait to be answered. During the waiting period callers are played various MOH (Music On Hold) sound files.

In this chapter we will cover:



9.1 Queues

This screen lists all system queues with the following details:



Name:
Queue name
Example: Patience
Field Type: Display

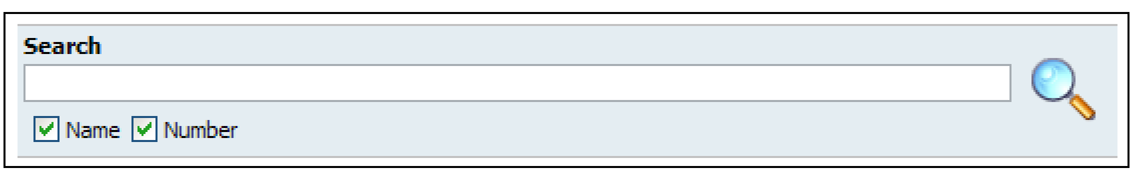
Number:
Queue network number
Example: 1001
Field Type: Display

Edits the queue configuration
Example: Click to edit queue configuration
Field Type: Button

Deletes a queue from the system
Example: Click to delete a queue from the system
Field Type: Button

9.1.1 Search

By selecting 'Search' Command, search menu will be displayed. Searches can be done by Name and Number



Search:

Search phrase

Example: Provide a search phrase here and hit enter to filter the records
Field Type: [a-z][0-9]

Name:

Should search filter be applied to queue names

Example: Check the box to search queue names
Field Type: Check box

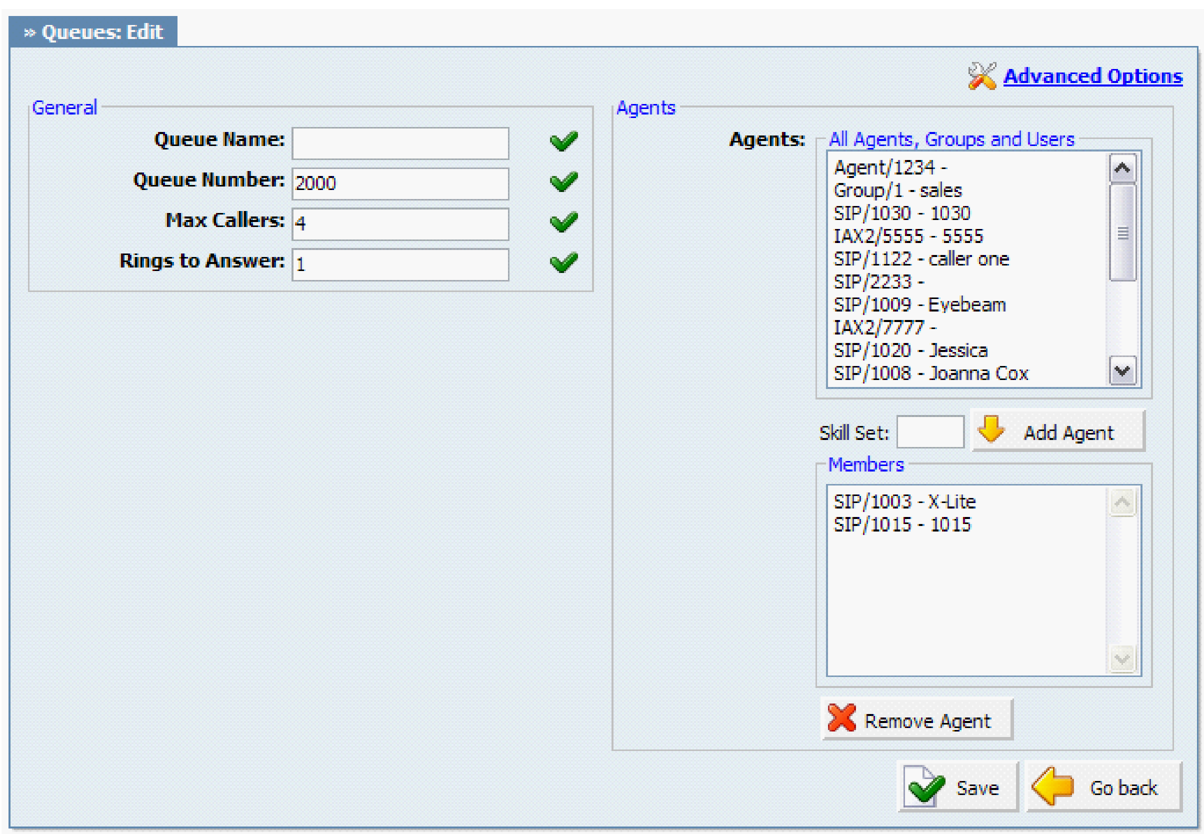
Number:

Should search filter be applied to queue numbers

Example: Check the box to search queue numbers
Field Type: Check box

9.1.2 Add/Edit IVR

Clicking on 'Add/Edit' Queue will open standard options shown below



Queue Name:

Unique queue network name/identifier

Example: Provide a unique queue identifier/name here
Field Type: [a-z][0-9]

Queue Number:

Unique network queue number

Example: This number is to be dialed in order to access the queue
Field Type: [0-9]

Max Callers:

Maximum number of callers allowed to wait in a queue at the same time. This number should be set in accordance to number of agents answering the queue calls

Example: If this field is set to 4, only 4 callers will be allowed to enter the queue. If caller number 5 tries to enter the queue he will be transferred to PBXware 'Redirect Extension' number.
Field Type: [0-9]

Rings to Answer:

Number of rings played to caller before entering the queue. Keep the rings between 1 and 3.

Example: Rather then just 'falling' into a queue and hearing the queue greeting message, play the ringing sound to caller first.
Field Type: [0-9]

Agents:

A list of all PBXware agents that can act as a queue members (e.g. SIP/1001 - John Doe).

Example: From this list select the queue agent and press the 'Add Agent' button. This will add selected agent to queue member list.
Field Type: Select box

Skill Set:

Prioritize which members are called by the system more often. Higher the skill equals less calls assigned to the agent

Example: If agent X is more skilled then agent Y, naturally, you would like to pass more calls to agent X. Set this field for agent X to 1, and for agent Y to 2. Once a skill is set for an agent, you can see it in the members list separated by a comma(SIP/1001 - John Doe,1)
Field Type: [0-9]

Add Agent:

Assigns one of the available agents to a queue members list

Example: If you would like a local user, John Doe, to answer the queue calls, select his name from the agents list and click this button. John Doe will be assigned to queue members list
Field Type: Button

Members:

A list of all members answering the queue calls

Example: If user John Doe is assigned to answer the queue calls, with all calls trying to go to him first(skill set = 1), 'SIP/1001 - John Doe,1' is displayed among other users/extensions, agents and agent groups
Field Type: Select box

Remove Agent:

Removes user, agent or agent group from the queue member list

Example: If user John Doe is no longer to answer the queue calls, just select his name from the members list and click his button
Field Type: Button

NOTE:

Static Login: Agent can login/logout from his extension only and stay logged in until the static logout number is dialed by the Agent.
Dynamic Login: Agent can login from any extension and stay logged in as long as the connection is not hangup by the Agent.
Dynamic Callback Login: Agent can login from any extension and stay logged in until the dynamic callback logout number is dialed by the agent.

To do a static login dial:
***200+$QUEUE_NUMBER** (*2003001)

To do a static logout dial:
***201+$QUEUE_NUMBER** (*2013001)

To do a dynamic login dial:
***202 + $AGENT_NUMBER + AGENT_PIN** (*2023001 + 1050)

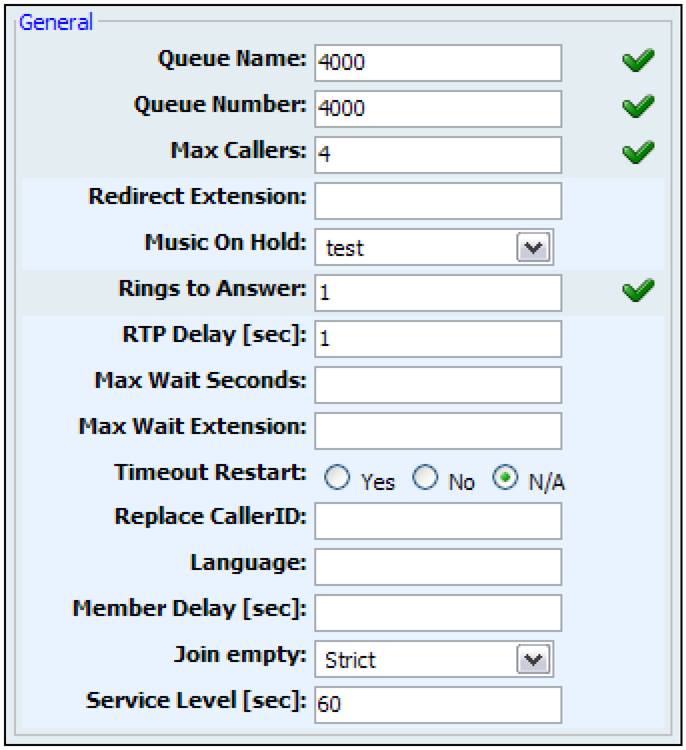
Once a caller is in a queue, agent can set any proper, mobile or network number to dynamically login into queue by dialing:
To do a dynamic callback login do:
***203 + $AGENT_NUMBER + AGENT_PIN + EXTENSION**(*2033001 + 1050)

To do a dynamic callback logout do:
***203 + $AGENT_NUMBER + AGENT_PIN + #**(*2033001)

9.1.3 Advanced Options

In this chapter we will cover:

9.1.3.1 General



Redirect Extension:

Max callers redirect extension number

Example: If queue 2000 accepts maximum of 4 users waiting at the same time, any new user that enters queue 2000 will be redirected to extension number provided here
Field Type: [0-9]

Music On Hold:

Select MOH(Music On Hold) class name. All sound files belonging to this MOH class will be played to users in queue

Example: User A enters the queue. After the greeting message is heard, all sound files belonging to selected MOH class are played in the background
Field Type: Select box

RTP Delay [sec]:

Delay time in seconds inserted before the queue greeting message is played. This solves the 'half-played' file problem. Keep this value between 1-3

Example: User A enters the queue and hears '..r call is first in line...' and wonders what was that.. what did the voice say!? Set this field to 1 so that 1 second pause is added before the message is played. Now, when user A enters the queue he will hear 'Your call is first in line...'.
Field Type: [0-9]

Max Wait Seconds:

Maximum time a caller can wait in a queue. Once this time is exceeded caller will be redirected to 'Max Wait Extension' number.

Example: User A is waiting 5 minutes already in the queue. If this field is set to 300(300s = 5min), A will be redirected to 'Max Wait Extension' so he doesn't loose his mind waiting :)
Field Type: [0-9]

Max Wait Extension:

This option works along with the 'Max Wait Seconds' field. Provide the extension to which caller will be redirected once time set under 'Max Wait Seconds' exceeds.

Example: User A is waiting 5 minutes already in the queue. If 'Max Wait Seconds' field is set to 300(300s = 5min), A will be redirected to this extension so he doesn't loose his mind waiting :)
Field Type: [0-9]

Timeout Restart:

Reset the internal timer if BUSY or CONGESTION is received from agent

Example: A call enters the queue and is transferred to Agent X. Agent X has a Polycom phone(for example) and send a BUSY signal by hitting the 'Reject' key. This will reset the internal timer. The call will be transferred to other queue agent(s). The original agent will not be contacted until all other agents are tried and the call does not get answered by any of them.
Field Type: Options buttons

Replace Caller ID:

Replace the caller id with the custom value

Example: Type 'Lobby - %CALLERID%' to display the caller id information as 'Lobby - 5552879' where 5552879 is the actual number calling in
Field Type: [a-z][0-9]

Language:

Define custom language for all sound files played by the queue

Example: To play Spanish sound files to all users waiting in the queue type 'es' here. NOTE: PBXware comes with English sound files by default. To install sound files in other languages please see ' Settings: Protocols: Sip: Language' for more information
Field Type: [a-z]

Member Delay:

This field is the same as RTP Delay, only this option is set for agent answering the queue calls. Before the call is transferred, custom queue information can be played to an agent so that agent knows from which queue the call is coming from. This solves the 'half-played' file problem. Keep this value between 1-3

Example: Agent X is to answer the call coming from the queue. If the 'Queue Announce' is set to play custom sound file('This call comes from the Lobby Queue') but only '...s call comes...' is heard, set this field to 1 so that 1 second pause is added before the message is played and entire message is played 'This call comes from...'
Field Type: [0-9]

Join Empty:

Set whether a caller can join a queue if no agent is logged in or unavailable

Example: It is recommended to set this option to 'No'. Do not allow user to enter the queue if the call will not be answered by anyone. Following options are available:
- Yes - Join queue if no agents or only unavailable agents are in the queue
- No - Do not join queue if no agents available
- Strict - Do not join queue in no agents or only unavailable agents are in the queue

Field Type: Option buttons

Service Level [sec]:

This options is used for the service statistics.

Used for service level statistics (calls answered within service level time frame)

Example: It is recommended to set this option to 'No'. Do not allow user to enter the queue if the call will not be answered by anyone. Following options are available:
- Yes - Join queue if no agents or only unavailable agents are in the queue
- No - Do not join queue if no agents available
- Strict - Do not join queue in no agents or only unavailable agents are in the queue

Field Type: Option buttons

Autofill:

Should callers be served one by one or in parallel fashion

Example: With this option turned 'Off', even if there are five agents available calls will not be transferred to them until first caller waiting in a queue is connected to an agent. When first caller gets served, caller number two gets served and all other keep waiting. Obviously it is recommended to keep this feature always turned 'On' so callers can be served in parallel
Field Type: Option buttons

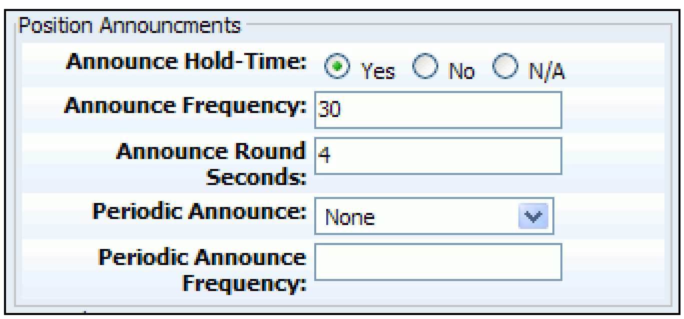
Ring Agents in Use:

Should agents in use be rang when new caller comes into queue

Example: If agent is already in active conversation, with this option set to 'Yes' Agent extension will ring when new caller enters the queue
Field Type: Option buttons

9.1.3.2 Position Announcements

There are two types of position announcements: 'Hold Time' (Tells the queue position) and 'Periodic Announcements' (Plays custom message)



Announce Hold-Time:

Enable callers waiting in a queue to hear the hold-time announcements.

Example: Setting this option to 'Yes' will enable the hold-time announcements. A single caller waiting in a queue would hear 'Your call is now first in line and will be answered by the next available representative. Thank you for your patience' message.
Field Type: Option buttons

Announce Frequency:

How often to play the hold-time announcement message (time in seconds). NOTE: If you set this option to '0', announce message will not be played.

Example: If this field is set to 30, a single caller waiting in queue will hear 'Your call is now first in line and will be answered by the next available representative. Thank you for your patience' message every 30 seconds.
Field Type: [0-9]

Announce Round Seconds:

This feature rounds announcement minutes and seconds to specific format.

Example: -
Field Type: [0-9]

Periodic Announce:

Select the sound file that is played periodically to callers waiting in a queue
NOTE: File name has to be in the following format 'periodic-announce-$NAME.gsm'

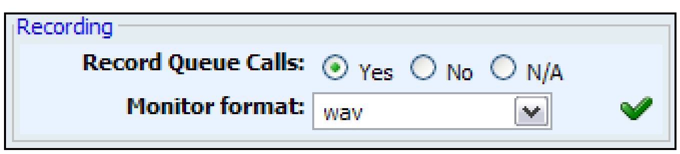
Example: Record a message 'Hang in there buddy!' and set it as a periodic announcement. This message will be played to callers every 'Periodic Announce Frequency' seconds
Field Type: Select box

Periodic Announce Frequency:

Time interval in seconds at which the periodic message is to be played.

Example: If this field is set to 30, all callers waiting in queue will hear the 'Periodic Announce' message every 30 seconds, as long as they stay in the queue.
Field Type: [0-9]

9.1.3.3 Recording



Record Queue Calls:

Once this feature is activated, all queues calls will be recorded in desired sound format.

Example: John enters the 'Sales' queue and is transferred to 'Queue Agent Smith'. Their entire conversation is recorded and available for review from 'Reports'.

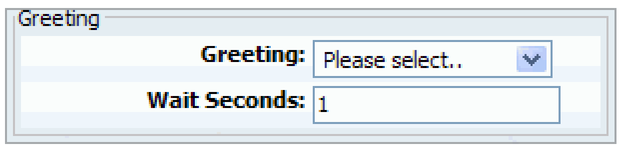
Field Type: Option buttons

Monitor format:

Select the audio format all queue calls with be recorded in. Available formats: gsm, wav and wav49.

Example: John enters the 'Sales' queue and is transferred to 'Queue Agent Smith'. Their entire conversation is recorded and available for review from 'Reports'.
Field Type: Option buttons

9.1.3.4 Greeting



Greeting:

Select a greeting file played to all callers waiting in a queue. NOTE: File name has to be in the following format 'queue-greeting-$NAME.gsm':

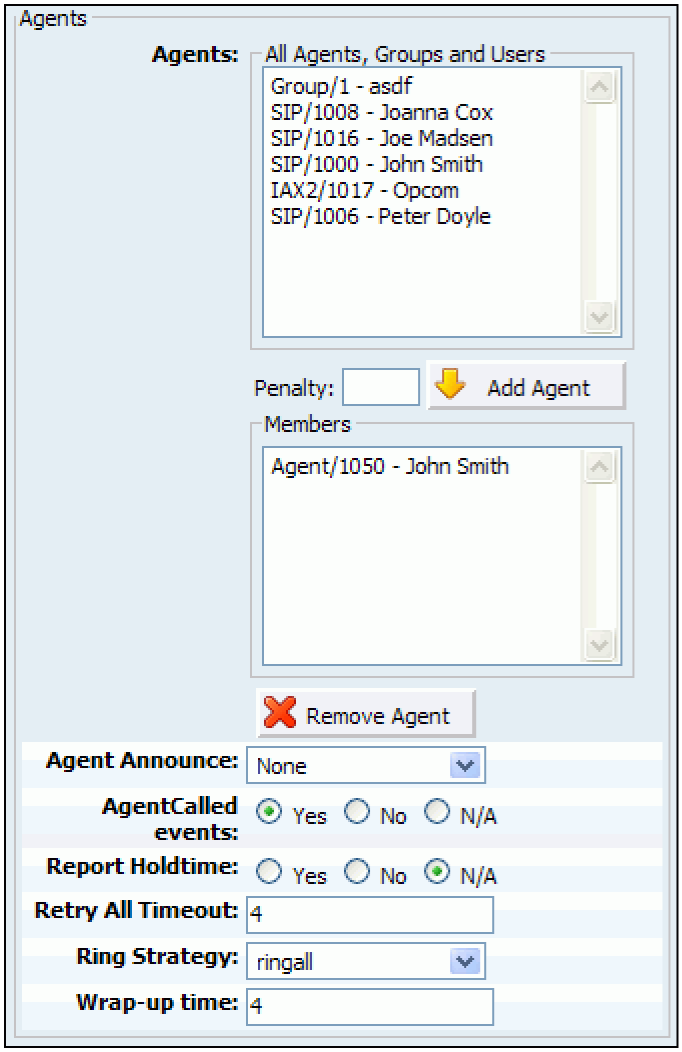
Example: Record a custom sound greeting file (e.g. 'All our representatives are busy...') and select that file. This file will be played to all callers once they enter the queue.
Field Type: Select box

Wait Seconds:

Delay time in seconds inserted before playing the greeting message. This delay is useful when users do not hear the beginning of the greeting message. Keep this value between 1-3 seconds.

Example: John enters the sales queue, but cannot hear the beginning of a greeting sound file file (e.g. ..ur representatives are busy...). Set this option to 1. This will insert one second of silence before the greeting file is played and should fix the partial sound file error. Now, all callers entering the queue should hear the full greeting message (e.g. All our representatives are busy...)
Field Type: [0-9]

9.1.3.5 Agents



Agent Announce:

Play a custom message to a queue agent before they answer the call. NOTE: File name has to be in the following format 'agent-announce-$NAME.gsm'.

Example: Record a custom sound file (e.g. This call is coming from Sales Queue). Every time queue agent answers the call from this queue, selected sound file will be played to him before the calls is transferred to agent
Field Type: Select box

Agent Called Events:

PBXware can generate some events when agent/member is connected. Enabling this feature may generate a large amount of extra manager events.

Example: It is recommended to always keep this feature set to 'Yes'
Field Type: Option buttons

Report Holdtime:

Enabling this feature will tell queue agent how long the calling party has been waiting in a queue.

Example: John is waiting in a sales queue for two minutes and thirty seconds. Agent Smith picks up his call and hears the message 'Hold time, two minutes'.
Field Type: Option buttons

Retry All Timeout:

Time interval in seconds, for how long to wait before trying queue agent again.

Example: If agent Smith cannot answer the incoming queue call and hangs up the ringing line, the call will not be transferred back to his extension for a time in seconds provided in this field.
Field Type: [0-9]

Ring Strategy:

Set the way calls are transferred to queue members answering the calls. Available options:
- ringall - Ring all available Members until one answers (default)
- roundrobin - Ringing each available Member
- leastrecent - Ring Member with least Queue calls
- fewestcalls - Ring Member with fewest completed Queue calls
- random - Ring random Member
- rrmemory - Round robin with memory. Remember where we left off last ring pass

Example: Read above for description
Field Type: Select box

Wrap-up Time:

After a successful call, this will determine how long to wait (time interval in seconds) before sending new call to potentially free agent/member.

Example:
Field Type: [0-9]

Queue members:

Any queue can have 3 type of queue members. These members can be: extensions, dynamic agents or call back agents

1. Extension
This member will receive queue calls upon loging into the queue by dialing *200$QUEUENUMBER.
For example: *2003000 (*200 access code and 3000 is queue number. From then on queue will be sending the calls to this member.

PLEASE NOTE:
In order for static agent to be able to login into a queue, an extenision needs to be a member of the queue in question.
Extension queue member must logout with *202, otherwise the queue will be sending the calls to it forever.

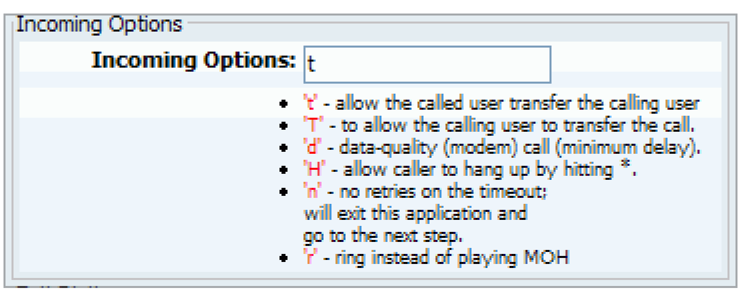
2. Dynamic Agents
This type of queue member can login into deignated queue from any UAD on the network with *202.
The system will prompt for AGENT NUMBER and AGENT PIN. From then on queue will be sending the calls to the agent.

PLEASE NOTE:
Dynamic agent is "on call" all the time. In another words.. The agents line is always open and receiving calls. Every time the caller hungs up, the agent hears the MOH and then next call will start. Dynamic egent
logs out of the queue just by hunging up the line.
The agent MUST be created in in the main menu "Agents" and assigned to the queue in the question.

3. Call Back Agents
This type of agent can login into designated queue by dialling into one of the system extensions with *203.
The system will prompt for:AGENT NUMBER, AGENT PIN, THE NUMBER (that agent is at currently at)
This number could be a local extension or some other remote destination.
For example: Extension: 1050 , Remote destionaton (032 345 233, agents home phone number)

PLEASE NOTE:
The agent MUST be created in in the main menu "Agents" and assigned to the queue in the question.
Call back agent must logout with *203, otherwise the queue will be sending the calls to it forever

9.1.3.6 Incoming Options

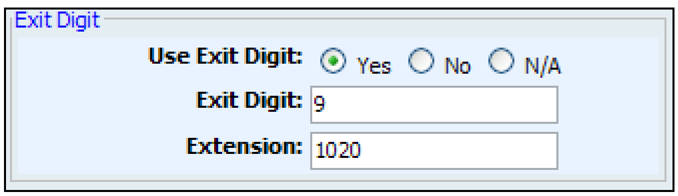


Incoming Options:

Set the advanced queue call options. Available options:
- t - allow the called user to transfer the calling user
- T - allow the calling user to transfer the call
- d - data-quality (modem) call (minimum delay)
- H - allow caller to hang up by hitting *
- n - no retries on the timeout; will exit queues and go to the next step
- r - ring instead of playing MOH

Example: Read above for description
Field Type: [a-z]

9.1.3.7 Exit Digit



Use Exit Digit:

Should users be able to exit the queue by dialing a single digit and be automatically redirected to a preset PBXware destination.

Example: John enters the sales queue. Greeting message explains that user may quit the queue by pressing digit defined under 'Exit Digit' option, and be transferred directly to operator or some other destination (set under 'Extension' option).
Field Type: Options buttons

Exit Digit:

Define the exit digit here. Once this digit is dialed by user waiting in queue, it will transfer the call directly to 'Extension' destination

Example: John enters the sales queue. Greeting message explains that user may quit the queue by pressing digit defined here (e.g. 9) and is transferred directly to operator or some other destination (set under 'Extension' option).
Field Type: [0-9]

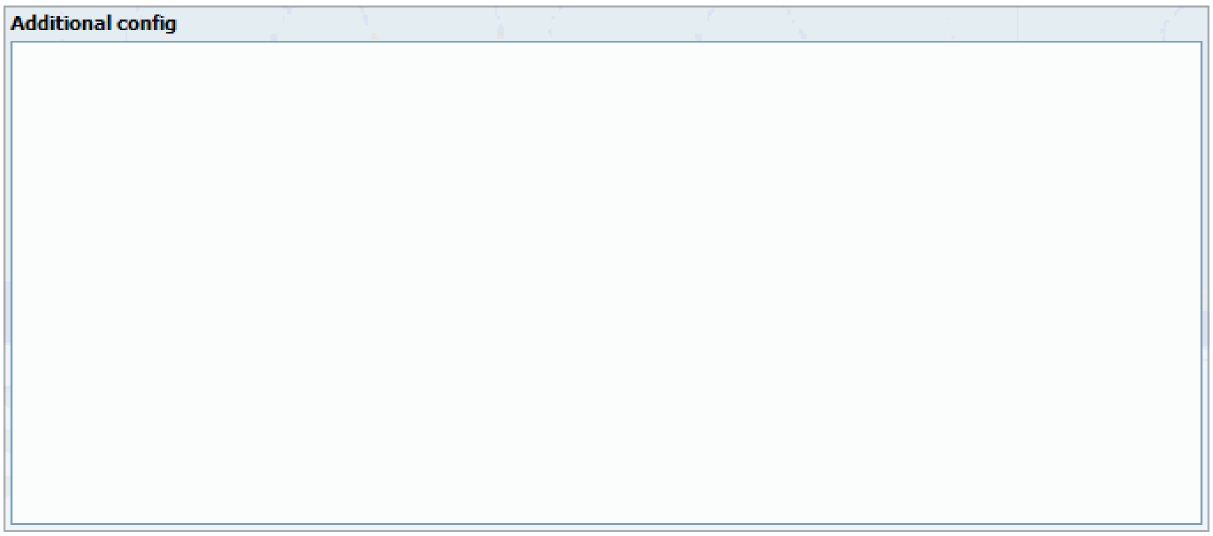
Extension:

Local PBXware extension number that is dialed once the 'Exit Digit' is dialed.

Example: John enters the sales queue. Greeting message explains that user may quit the queue by pressing digit defined under 'Exit Digit' option, and be transferred directly to PBXware extension defined here.
Field Type: [0-9]

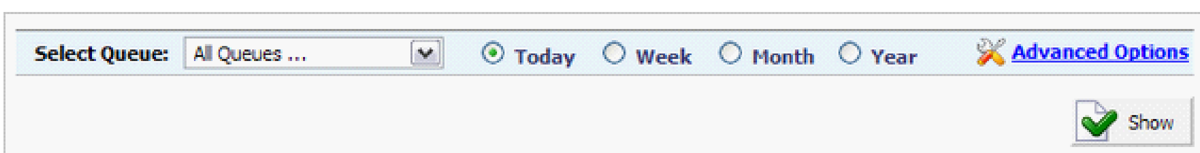
9.1.3.8 Additional Config

In this section additional information can be added to be written into the queue configuration file.



9.2 Statistics

Queue statistics display a wide and detail array of queue details



Select Queue::

Select the queue you wish to see the statistics for.

Example: Select 'All Queues' or a specific queue name here and click the 'Show' button
Field Type: Select box

Today/Week/Month/Year:

Statistics time period

Example: Select the time period statistics is to be displayed for. Available options:
- Today
- Week
- Month
- Year

Field Type: Option buttons

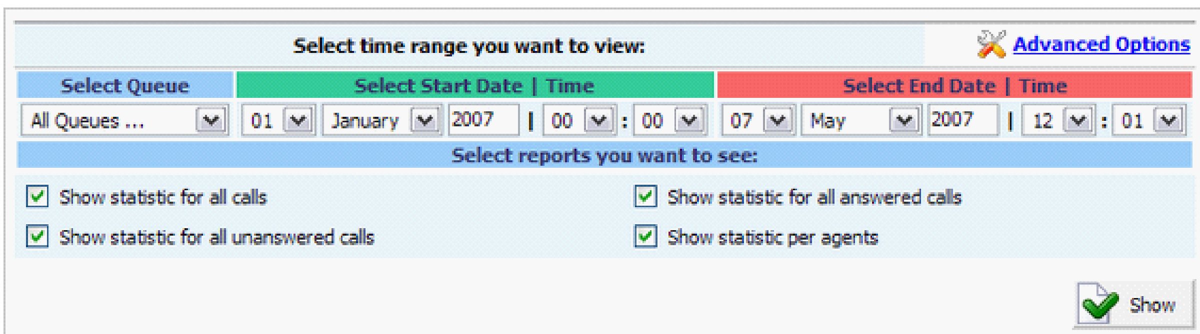
Show:

Display queue statistics

Example: A click on this button will open the statistics popup window. Displayed information may vary depending on the options set under basic and advanced options.
Field Type: Action Button

9.2.1 Advanced Options

Advanced Options:
To access advanced interface, click on 'Advanced Options' located in the top right corner.



These options set the time range and specific query summaries:

- **Show statistic for all calls** – All incoming calls summary
- **Show statistic for all answered calls** – All incoming calls answered by the agents
- **Show statistic for all unanswered calls** – All unanswered calls
- **Show statistic per agents** – Per queue agent call statistics

9.2.2 Report Window

Queues statistic report window

General Actions:

- **Download Report** – If you choose this action PBXware will generate a pdf format report and offer it for download or view.
- **Print** – This will print statistic report on yours printer.
- **Email** – This action will generate pdf report attach it to message and offer you to email it.
- **Close Window** – This will close report window.

Call Actions:

- **More** – This will print statistic report on yours printer.
- **Graph** – This action will generate pdf report attach it to message and offer you to email it.
- **Get CSV** – This will close report window.

All/Answered/Unanswered/Agent Calls:

- **Table caption** - Displays the total number of all calls made
- **Total** - Total call time including the hold time as well
- **Min** - The minimum/shortest call time
- **Mean** - Average call time for all incoming calls
- **Max** - Max call time.

- **Call Time** - Calls active talk time in a queue
- **Hold Time -** Calls hold time in a queue
- **Entry Position** - Caller position in a queue at a point of entry(Total value always displayed as '-')

- **Agent Hangups** – Number of calls ended by queue agents and their percentage compared to all unanswered calls.
- **Caller Hangups** – Number of calls ended by callers and their percentage compared to all unanswered calls.
- **Service Level** – Percentage of calls answered in the first 60 seconds. (This time period can be changed in queue properties).

NOTE:
Answered and Unanswered summary displays SLA(Service Level Agreement) data in percents taken in 10 seconds interval.

Agent Actions:
- Sessions
- Agent/$NUMBER

Agents:
- **Calls** – All calls answered by agent
  - **Number** – Number of calls answered by agent
  - **Per min** – Number of calls per minute
  - **Percent** – Percent of all answered calls by the agent

- **Talk Time** – Agent talk time
  - **Total** – Total agent talk time
  - **Mean** – Mean/Average agent talk time
  - **Mean Delay** – Mean/Average caller wait time before answered by the agent

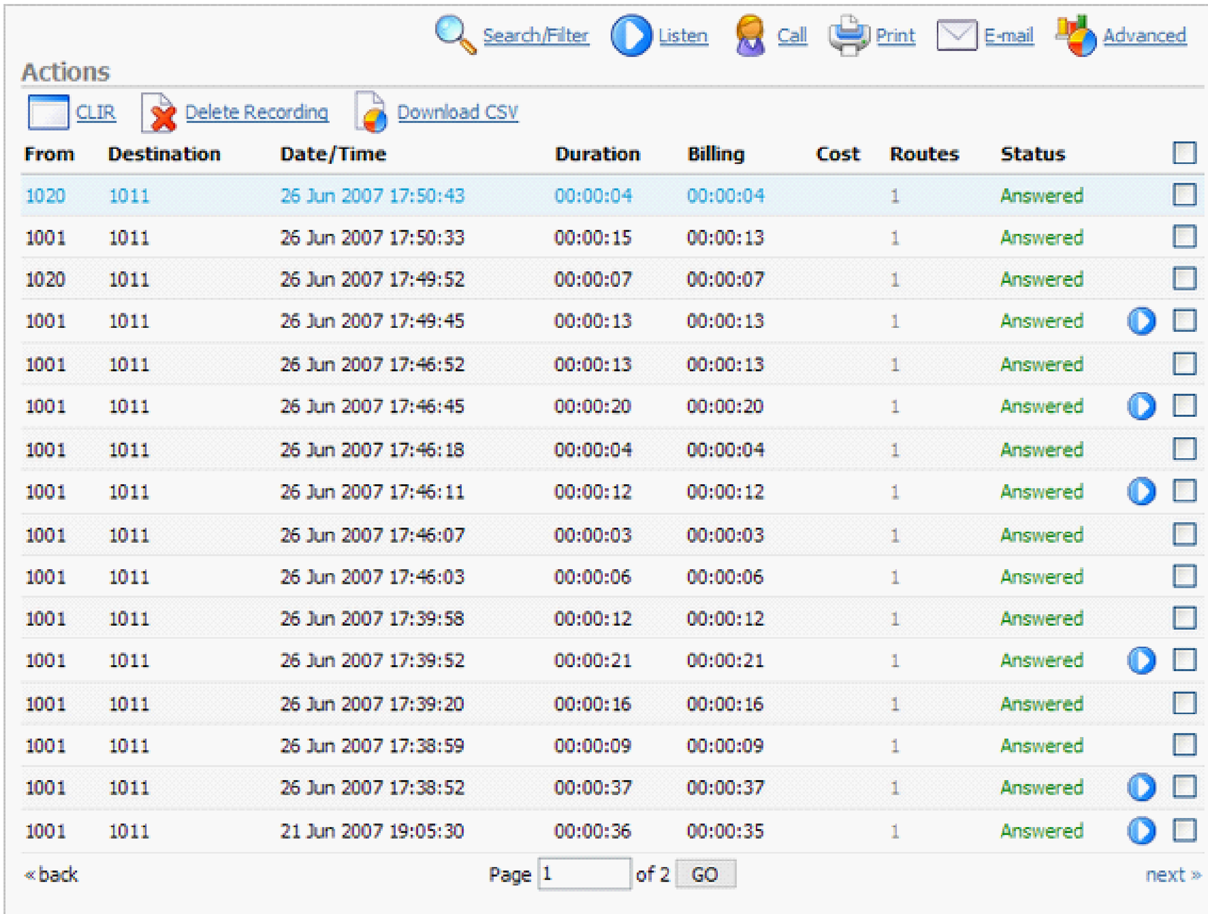
- **Idle Time** – Agent idle time (callback agents only)
  - **Total** – Total agent idle time(when agent was not busy)
  - **Mean** – Mean/Average idle time between calls

- **Session Time** – Time agent was logged in the queue (callback agents only)
  - **Total** – Total time agent was logged in
  - **Mean** – Mean/Average time agent was logged in
  - **Count** – Number of agent sessions.

- **Hangups (Agent/Caller)** – Hang-ups statistic for agents and callers
  - **Number** - Number of call hang-ups by Agent/Caller
  - **Percent** – Percentages of call hang-ups by Agent/Caller

9.3 CDR

CDR (Call Detail Records) for all placed or received calls on the system. In addition to normal operation an authorized user is able to perform additional actions such as extensive search, listen to recorded calls, call any destinations listed and access advanced features.



From	Destination	Date/Time	Duration	Billing	Cost	Routes	Status
1020	1011	26 Jun 2007 17:50:43	00:00:04	00:00:04		1	Answered
1001	1011	26 Jun 2007 17:50:33	00:00:15	00:00:13		1	Answered
1020	1011	26 Jun 2007 17:49:52	00:00:07	00:00:07		1	Answered
1001	1011	26 Jun 2007 17:49:45	00:00:13	00:00:13		1	Answered
1001	1011	26 Jun 2007 17:46:52	00:00:13	00:00:13		1	Answered
1001	1011	26 Jun 2007 17:46:45	00:00:20	00:00:20		1	Answered
1001	1011	26 Jun 2007 17:46:18	00:00:04	00:00:04		1	Answered
1001	1011	26 Jun 2007 17:46:11	00:00:12	00:00:12		1	Answered
1001	1011	26 Jun 2007 17:46:07	00:00:03	00:00:03		1	Answered
1001	1011	26 Jun 2007 17:46:03	00:00:06	00:00:06		1	Answered
1001	1011	26 Jun 2007 17:39:58	00:00:12	00:00:12		1	Answered
1001	1011	26 Jun 2007 17:39:52	00:00:21	00:00:21		1	Answered
1001	1011	26 Jun 2007 17:39:20	00:00:16	00:00:16		1	Answered
1001	1011	26 Jun 2007 17:38:59	00:00:09	00:00:09		1	Answered
1001	1011	26 Jun 2007 17:38:52	00:00:37	00:00:37		1	Answered
1001	1011	21 Jun 2007 19:05:30	00:00:36	00:00:35		1	Answered

« back Page 1 of 2 GO next »

From:
Extension number the call was made from
Example: If call was made from extension 1001 to extension 1004, '1001' is displayed here.
Field Type: Display

Destination:
Extension number the call was made to
Example: If call was made from extension 1001 to extension 1004, '1004' is displayed here.
Field Type: Display

Date/Time:
Date and Time when the call was made
Example: 04 Oct 2006 10:44:10

Field Type: Display

Duration:
Call duration time in hh:mm:ss format
Example: 00:12:45
Field Type: Display

Billing:
Time billed by the system
Example: 00:12:45
Field Type: Display

Cost:
Total cost of the call calculated through a service plan
Example: 0.71
Field Type: Display

Routes:
Number of system routes
Example: If call goes directly from extension to destination - '1' will be displayed here. If call has entered the IVR and was then redirected to Queue and then Agent answers the call '2' will be displayed here (IVR and Queue were used)
Field Type: Display

Status:
Displays the call status
Example: Depending on whether a call was answered or not, this field value may have the following content:

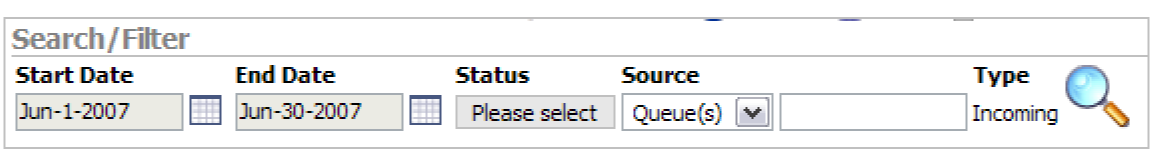
- Answered
- Not Answered
- Busy
- Error

Field Type: Display


This icon is displayed once a call is recorded and 'Delete' or 'Listen' enhanced service is active


This is a box used with the CDR commands to select a desired call

9.3.1 Search/Filter



Start Date:

Select a Search/Filter start date

Example: Click on a small 'Calendar' icon next to a field and select desired date

Field Type: Option button

End Date:

Select a Search/Filter end date

Example: Click on a small 'Calendar' icon next to a field and select desired date
Field Type: Option button

Status:

Search calls by selecting desired call status

Example: Click on a 'Please Select' button and select one of the available fields:
- All
- Answered
- Not Answered
- Busy
- Error

Field Type: Select box

Source:

Search calls by the source they arrived

Example: Selecting 'Queue' and typing a queue number into a text field next to this (e.g. 1011) will display all calls that came over queue 1001.
Field Type: Option button

Type:

Search calls based on the type of calls

Example: Click the 'Type' button and select one of the available fields:
- All
- Outgoing
- Incoming

Field Type: Select box

NOTE: After making any changes to search filter, be sure to click the  search icon

9.3.2 Actions

In this chapter we will cover:

9.3.2.1 Listen

Listen:

Once the 'Listen' icon is displayed next to a call record it means that the specific call was recorded.

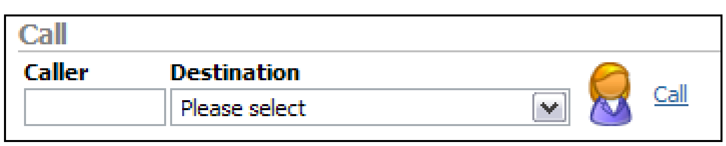
Example: To play recorded calls, check the box next to a 'Listen' icon and click 'Listen'. Browser will prompt you to open the sound file in your favorite audio player or to download the sound file.
Field Type: Option button

NOTE: By default the sound format is available as a .gsm file. To change the recording format go to: 'Settings: Servers: Edit: Recordings format' and select one of the available sound formats:

- gsm
- wav
- wav49 and
- ogg

9.3.2.2 Call

To establish a call between any PBXware extension with a listed extensions you have to provide only two things. The Caller $EXTENSION number and the $DESTINATION extension



Caller

PBXware extension that will make a call

Example: Provide any PBXware extension number here, 1001 for example

Field Type: [0-9]

Destination:

Destination extension that will be dialed by 'Caller' extension

Example: To select a destination extension, first check a box next to a CDR record. This field will display two extensions listed under 'From' and 'Destination' selected record
Field Type: Select button

NOTE: After setting 'Caller' and 'Destination' extensions click the  call icon

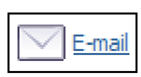
9.3.2.3 Print

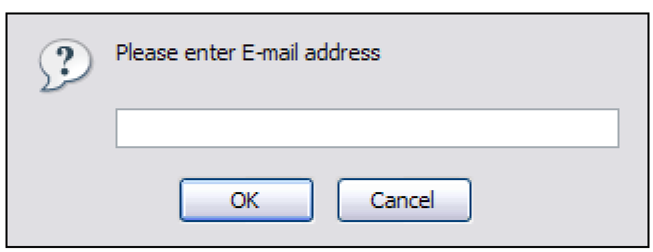
Check the box next to a call record and click the 'Print' button. This action will open a new popup window with the printing interface.

From	Destination	Date/Time	Duration	Status
5555	4000	30 Jan 2007 11:25:02	00:00:24	Answered
1000	4000	29 Jan 2007 17:21:20	00:00:14	Answered
1000	4000	29 Jan 2007 17:20:45	00:00:25	Answered
1000	4000	29 Jan 2007 17:03:13	00:00:27	Answered
1000	4000	29 Jan 2007 17:02:47	00:00:14	Answered
1000	4000	29 Jan 2007 16:59:52	00:00:24	Answered
1000	4000	29 Jan 2007 16:59:50	00:00:09	Answered
1000	4000	29 Jan 2007 16:59:49	00:00:00	Answered
1000	4000	29 Jan 2007 16:59:29	00:00:19	Answered
1000	4000	29 Jan 2007 16:59:20	00:00:27	Answered
1000	4000	29 Jan 2007 16:58:48	00:00:38	Answered
1000	4000	29 Jan 2007 16:58:45	00:00:33	Answered
1000	4000	29 Jan 2007 16:58:33	00:00:13	Answered
1000	4000	29 Jan 2007 16:58:22	00:00:21	Answered
1000	4000	29 Jan 2007 16:58:12	00:00:15	Answered
1000	4000	29 Jan 2007 16:57:50	00:00:19	Answered

9.3.2.4 Email

Check the box next to a call record and click the 'Email' button. A small popup dialog will appear. Provide email address here and click 'OK' button to send the records.

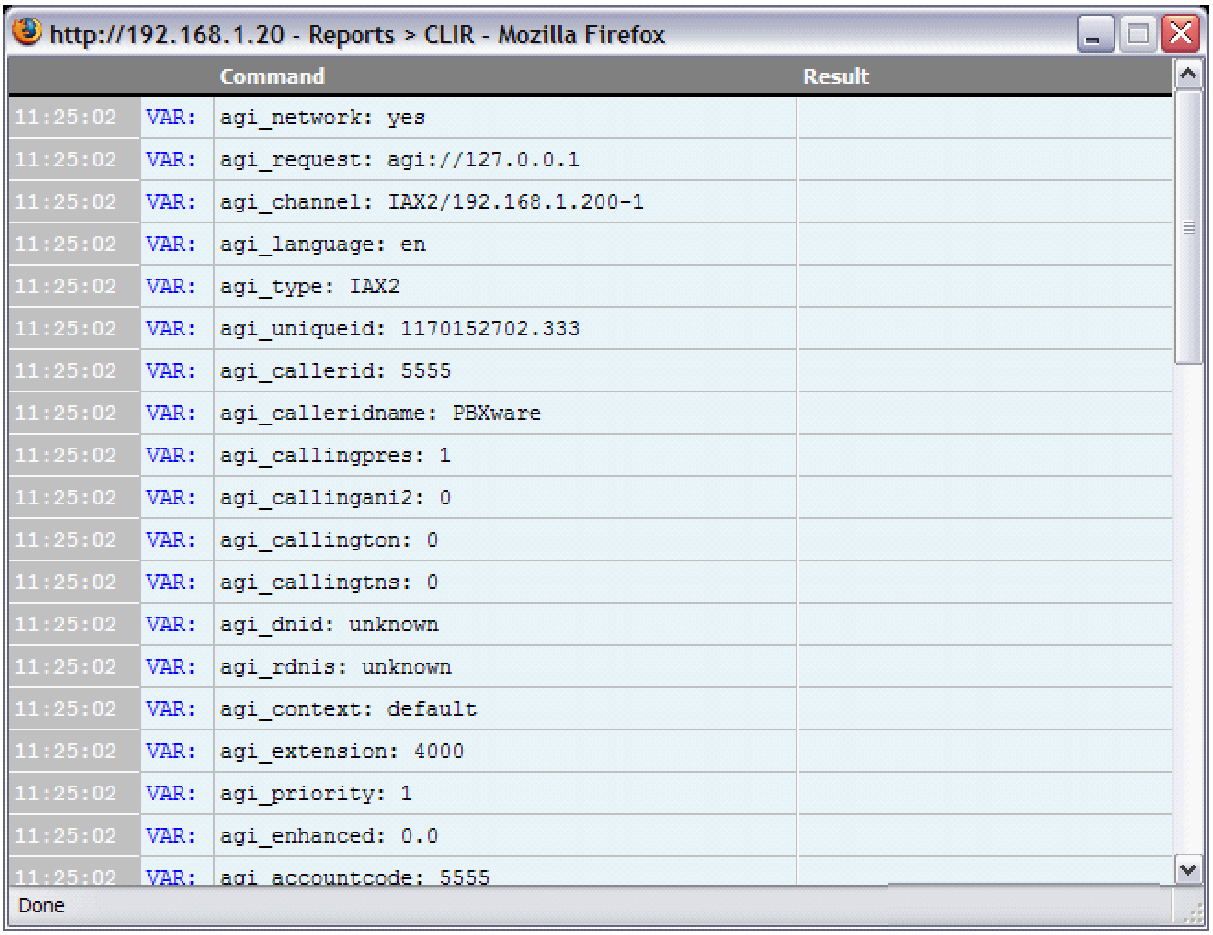




9.3.2.5 Advanced

In this chapter we will cover:
- CLIR 209
- Delete Recording 210
- Download CSV 210

9.3.2.5.1 CLIR



		Command	Result
11:25:02	VAR:	agi_network: yes	
11:25:02	VAR:	agi_request: agi://127.0.0.1	
11:25:02	VAR:	agi_channel: IAX2/192.168.1.200-1	
11:25:02	VAR:	agi_language: en	
11:25:02	VAR:	agi_type: IAX2	
11:25:02	VAR:	agi_uniqueid: 1170152702.333	
11:25:02	VAR:	agi_callerid: 5555	
11:25:02	VAR:	agi_calleridname: PBXware	
11:25:02	VAR:	agi_callingpres: 1	
11:25:02	VAR:	agi_callingani2: 0	
11:25:02	VAR:	agi_callington: 0	
11:25:02	VAR:	agi_callingtns: 0	
11:25:02	VAR:	agi_dnid: unknown	
11:25:02	VAR:	agi_rdnis: unknown	
11:25:02	VAR:	agi_context: default	
11:25:02	VAR:	agi_extension: 4000	
11:25:02	VAR:	agi_priority: 1	
11:25:02	VAR:	agi_enhanced: 0.0	
11:25:02	VAR:	agi_accountcode: 5555	

CLIR:

CLIR (Command Line Interface Record) details

Example: Select a desired call record and click this button to view more technical details about the call. A small popup window will open with the data. NOTE: When experiencing any kind of unexplained problems, this is the data you need to send to the technical support team
Field Type: Command Button

9.3.2.5.2 Delete Recording

Delete Recording:

Deletes the recorded calls. NOTE: For this command to be displayed, appropriate enhanced service has to be set.

Example: Select a recorded call and click this button to delete it from the file system
Field Type: Command Button

9.3.2.5.3 Download CSV

Download CSV:

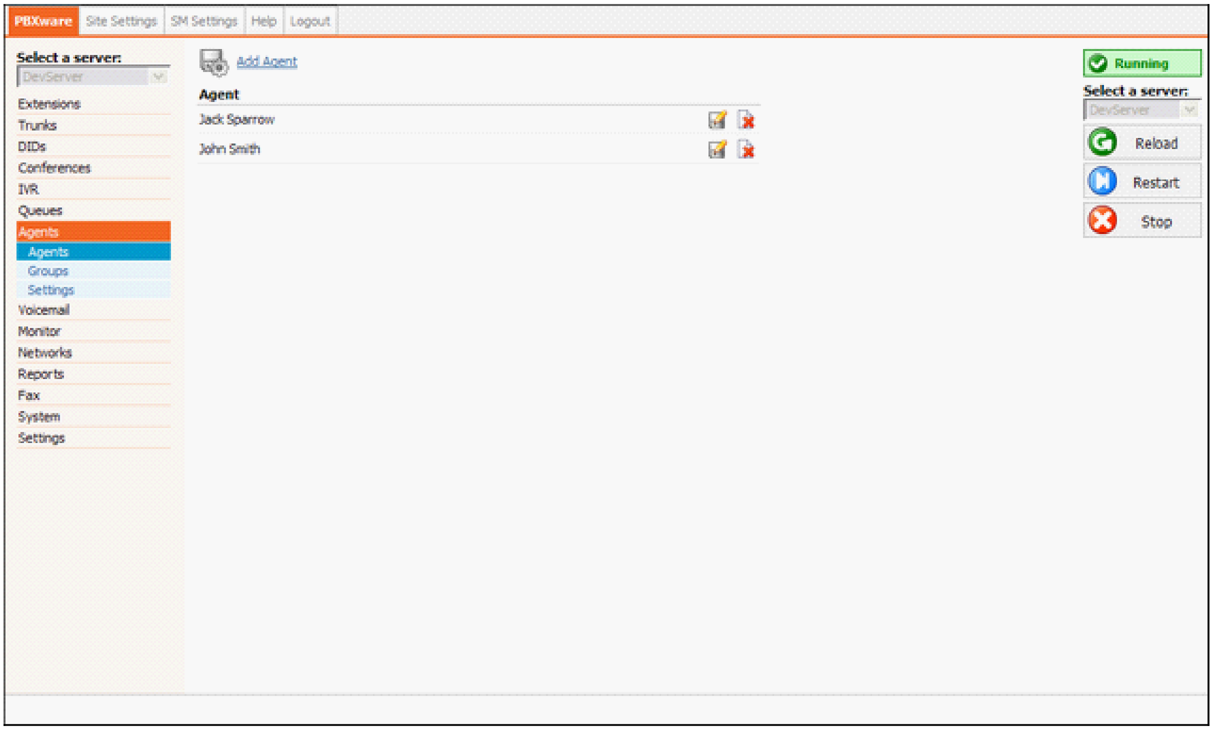
Download data as the .csv (Comma Separated Value) file

Example: Click this button to download the .csv file to your desktop
Field Type: Command Button

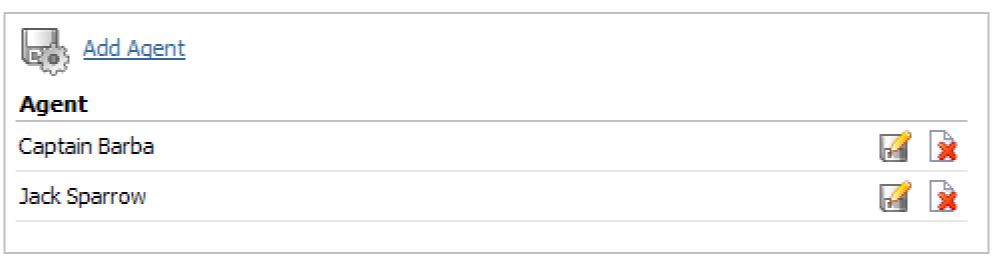
Part X

10 Agents

Queue agents are virtual system extensions. Agents login into queue member list and answer queue calls. Advantage of this system is that any system extension can login as an agent into a queue. This screen lists all system agents with the following details



10.1 Agents

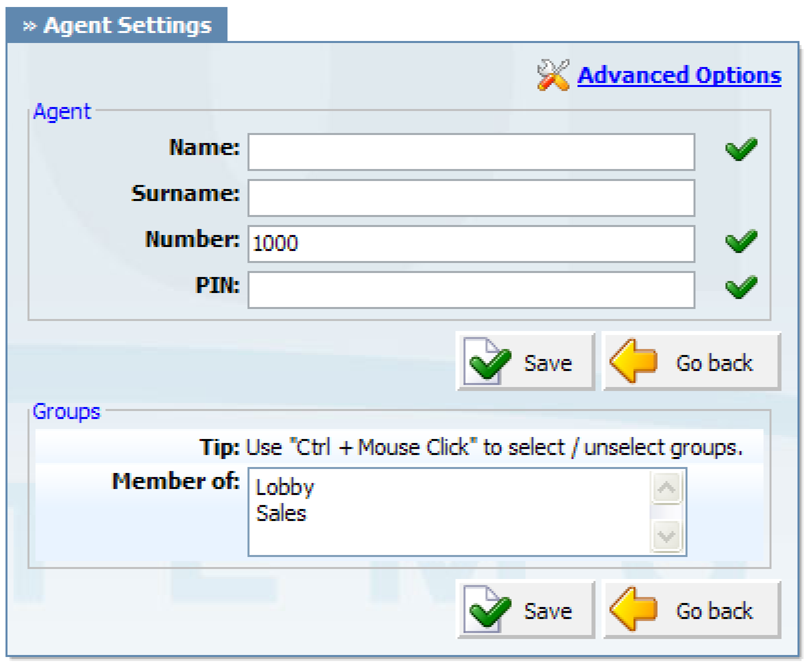


Agent:
Agent's full name
Example: Jack Sparrow
Field Type: Display

Edits Queue Agent configuration
Example: Click to edit Agent configuration
Field Type: Button

Deletes Queue Agent from the system
Example: Click to delete Queue Agent from the system
Field Type: Button

10.1.1 Add/Edit Agent



Name:

Queue agent name

Example: This name is used only for easier navigation. Set it to 'John' for example
Field Type: [a-z]

Surname:

Queue agent surname

Example: Example: This name is used only for easier navigation. Set it to 'Smith' for example

Field Type: [a-z]

Number:

Queue agent network number

Example: This number is provided by the agent when logging into queue. Set it to '4010' for example
Field Type: [0-9]

PIN:

Queue agent PIN number

Example: This number is provided by the agent when logging into queue. Set it to '6583' for example
Field Type: [0-9]

Member of:

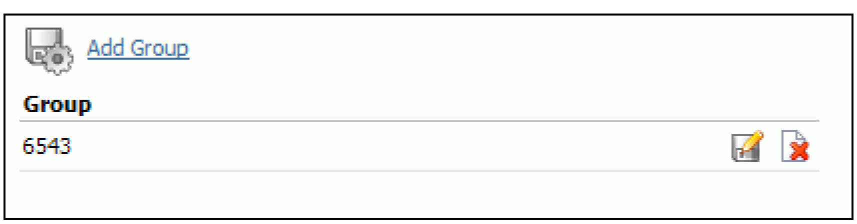
Groups queue agent belongs to

Example: By assigning agent to multiple groups, agent can login into multiple queues with a single agent login. Use CTRL + mouse click to select/unselect groups
Field Type: Select box

10.2 Groups

To enable easier login into multiple queues, agents can be organized in groups.

This way, many agents can be added to a queue member list with a single click (by adding agent group and not agents one by one). Also, when any system extension logs in as a queue member, it logs in automatically to all queues that agent is a member of.



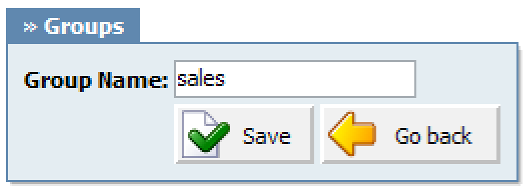
Group:

Queue group number
Example: 2000
Field Type: Display


Edits the queue group configuration
Example: Click to edit queue group configuration
Field Type: Button


Deletes a queue group from the system
Example: Click to delete a queue group from the system
Field Type: Button

10.2.1 Add/Edit Group



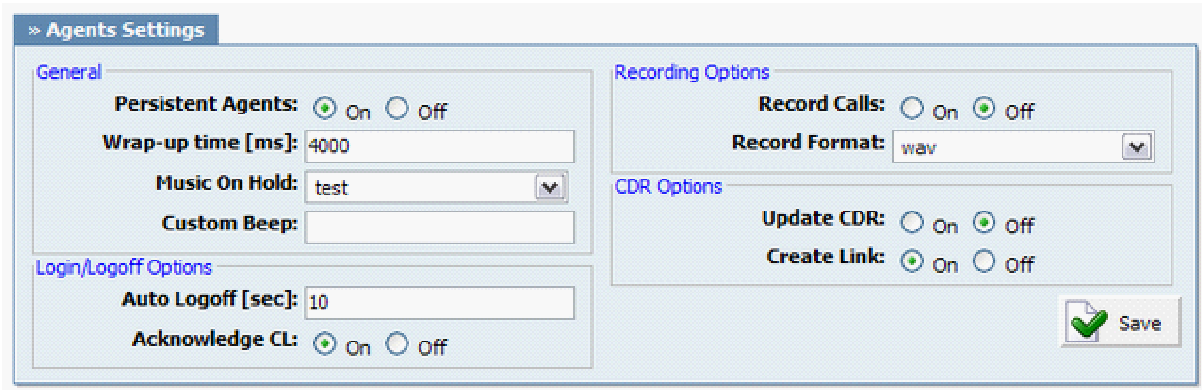
Group Name:

Unique network group identifier

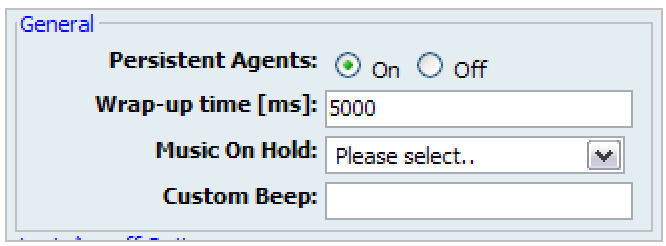
Example: Adding 'Sales' here will create 'Sales' group. Select this group under 'Agents: Member of' to assign the user to this group.
Field Type: [a-z] [0-9]

10.3 Settings

In this chapter we will cover:



10.3.1 General



Persistent Agents:

Should the callback login be restored once the PBXware restarts (The data is stored in database)

Example: If agent Smith is logged in sales queue, and PBXware needs to be restarted due to any reason, with this option set to 'Yes', agent Smith will be automatically registered with the sales queue after the system restart.
Field Type: Option buttons

Wrap-up time [ms]:

Pause in milliseconds given to queue agent before new call is transferred to his extension

Example: Agent Smith ends the queue call. A pause in ms (default 5000 = 5sec) is enforced so agent can prepare himself for the new call
Field Type: [0-9]

Music On Hold:

Set the Music On Hold class played to queue agents

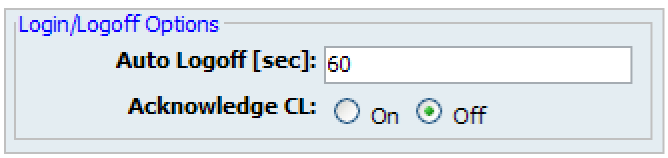
Example: Select 'Default' to play all sound files located in that class to queue agent while waiting for new calls
Field Type: Select box

Custom Beep:

Set the sound file name that is to be played to queue agent before a call is transferred to him from the queue

Example: Type 'arlington' to playe the arlington.gsm sound file to queue agent
Field Type: [a-z] [0-9]

10.3.2 Login/Logoff Options



Auto Logoff [sec]:

Time in seconds queue agent's extension is to ring before declaring him unavailable and logging him off the queue

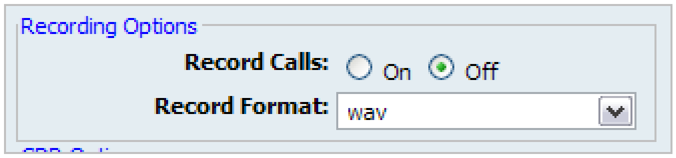
Example: If this field is set to 60, and queue agent doesn't answer the ringing extension during that time, he will automatically get logged off a queue so calls will be transferred to his extension until the next queue login
Field Type:

Acknowledge CL:

Require the agents that are logged in by agent callback login to dial # to confirm their input

Example: With this option set to 'Yes', once agent dials *203 he is asked to provide extension number and acknowledge it by dialling #. After correct extension, agent is asked for account password and and to acknowledge it by dialling #.
Field Type: Option button

10.3.3 Recording Options



Record Calls:

Record all calls made to queue agent

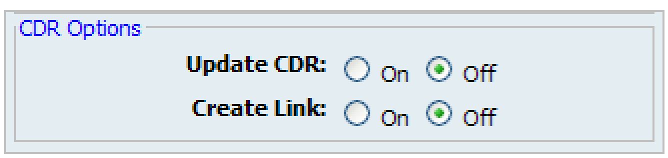
Example: With this option set to 'Yes', all calls agent receives through queue will be recorded
Field Type: Option button

Record Format:

Select desired sound format for call recording

Example: Select among available sound formats: wav, gsm, wav49
Field Type: Select box

10.3.4 CDR Options



Update CDR:

Should the CDR records be changed so it is known which agents generates the call

Example: -
Field Type: Option button

Create Link:

Should the CDR records display the name of the call recording

Example: -
Field Type: Option button

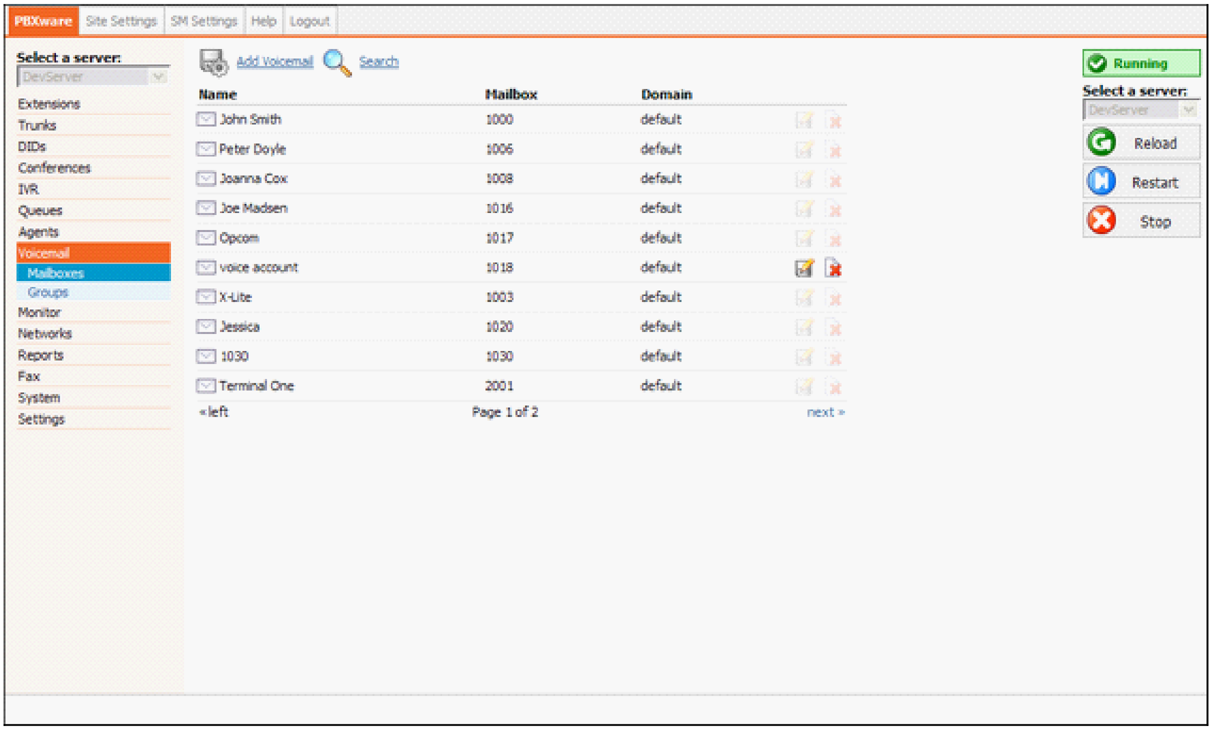
Part XI

11 Voicemail

PBXware voicemail is an advanced answering machine. Although each extension is equipped with a voice mailbox, voice mailboxes can be created on its own as well from this location.

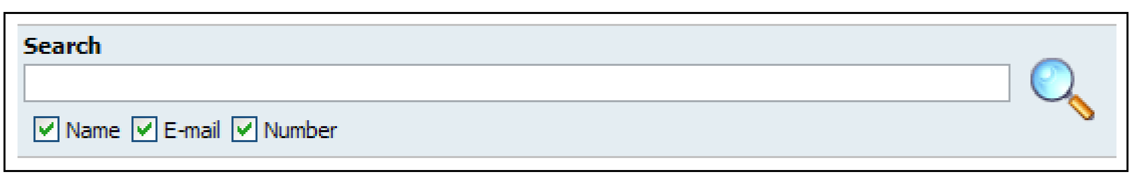
In this chapter we will cover:

- Search 221
- Add/Edit Voicemail 224
- Groups 229



11.1 Search

By selecting 'Search' Command, search menu will be displayed. Searches can be done by Name, Email and Extension number



Search:

Search phrase

Example: Depending on which check boxes are selected below (Name, E-mail, Number) provide corresponding phrase here. For example, if e-mail is selected bellow, type some email address here e.g. email@domain.com and click the search icon or hit enter on the keyboard
Field Type: [a-z][0-9]

Name:

Search voice mailboxes by user name

Example: Check this box and under 'Search' type the user's name or surname and click the Search icon or hit enter on the keyboard to display results
Field Type: Check box

E-mail:

Search voice mailboxes by email address

Example: Check this box and under 'Search' type the user's email address and click the Search icon or hit enter on the keyboard to display results
Field Type: Check box

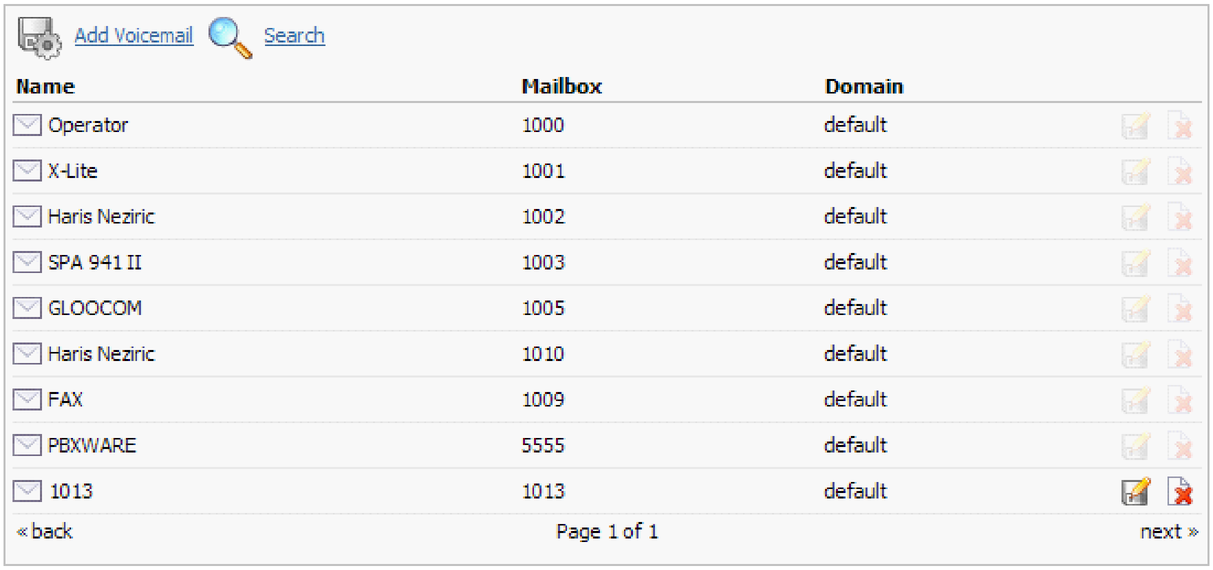
Number:

Search voice mailboxes by network extension

Example: Check this box and under 'Search' type the voice mailbox network extension and click the search icon or hit enter on the keyboard to display results
Field Type: Check box

11.2 Mailboxes

This screen lists all system mailboxes with the following details:



Name:
Full name of the voice mailbox user
Example: Peter Doyle
Field Type: Display

Mailbox:
Voice mailbox extension number
Example: 1006
Field Type: Display

Domain:
Domain/Context voice mailbox belongs to
Example: default
Field Type: Display


Edits the voice mailbox configuration
Example: Click to edit voice mailbox configuration
Field Type: Button


Deletes a voice mailbox account from the system
Example: Click to delete a voice mailbox from the system
Field Type: Button

NOTE: Edit and Delete commands will be disabled for users with the system extension. Their voicemail settings are edited via self care or by editing their extension.

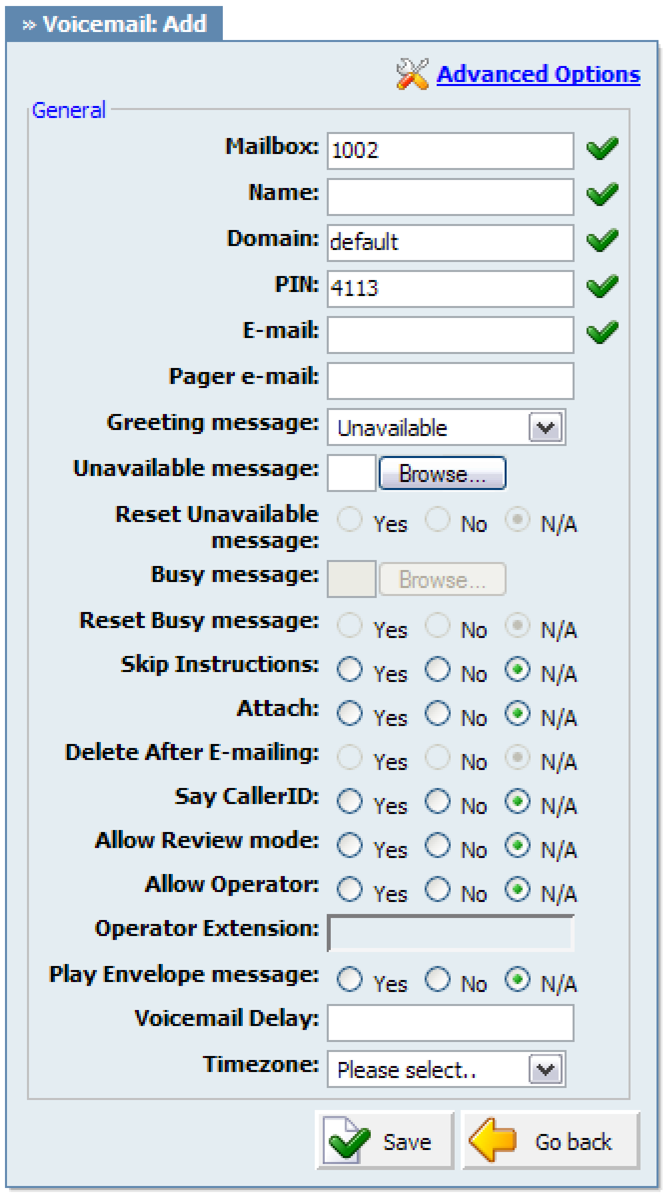
Voice Mailbox Access:

In order to access custom voice mailboxes from any PBXware extension dial '*124 +

$VOICE_MAILBOX_NUMBER'. For example '*124 2000',

11.2.1 Add/Edit Voicemail

Clicking on 'Add Voicemail/Edit' Voicemail will open voicemail screen shown below.



Mailbox:

Unique network voice mailbox extension number

Example: Set this field to 5001 for example. Now, in order to dial into this voice mailbox, simply dial 5001 from any PBXware extension
Field Type: [0-9]

Name:

Full name of voice mailbox owner

Example: John Smith
Field Type: [a-z]

Domain:

Domain/Context this voice mailbox belongs to

Example: Advanced feature. Leave this field set to 'default' unless you are know what you are doing
Field Type: [a-z][0-9]

PIN (Personal Identification Number):

Four digit number used for voice mailbox authentication

Example: Each voice mailbox has a unique PIN. In order to login to your voice mailbox, provide this number once asked for it by the operator e.g. 1947
Field Type: [0-9]

Email:

Email address associated with the voice inbox. This email is used for new voice message notification and audio file attachments

Example: If 'john@domain.com' is set here, once this mailbox receives a new message, notification and attached voice message (depending if this option is enabled) is sent to this email address
Field Type: [a-z][0-9][@_-.]

Pager e-mail:

Provide the pager e-mail address here

Example: If 'john@pager.com' is set here, once this mailbox receives a new message, notification is sent to this pager email address

Field Type: [a-z][0-9][@_-.]

Greeting message:

Greeting message played to uses before they are transferred to voice mailbox to leave the message

Example: Mailbox user may choose between 'Busy' and 'Unavailable' message
Field Type: Select box

Unavailable message:

Upload the custom unavailable message. NOTE: Unavailable message supports: WAV, wav and gsm files only

Example: If the default unavailable message does not suit you needs, click the 'Browse' button, upload a custom message and select it under 'greeting message'
Field Type: Select box

Reset Unavailable message:

Reset current unavailable message

Example: To reset current unavailable message click this button. NOTE: The message will be deleted from the filesystem
Field Type: Option buttons

Busy message:

Upload the custom busy message. NOTE: Busy message supports: WAV, wav and gsm files only

Example: If the default busy message does not suit your needs, click the 'Browser' button, upload a custom message and select it under 'greeting message'
Field Type: Select box

Reset Busy message:

Reset current busy message

Example: To reset current busy message click this button. NOTE: The message will be deleted from the filesystem
Field Type: Option buttons

Skip instructions:

Skip the instructions telling users how to leave a voice message

Example: Once the caller reaches the voice mailbox, instructions on how to leave voice message are played. You are encouraged to set this option to 'Yes' all the time
Field Type: Option buttons

Attach:

Should the voice message be attached and sent along with the notification email

Example: Caller leaves a voice message to John. With this option set to 'Yes', notification email John gets will have a voice message attached to it so John can listen to it without signing in to his voice mailbox
Field Type: Option buttons

Delete After E-mailing:

Should the voice message sound file be deleted form the filesystem after sending it as an attachment to user's email address

Example: Caller leaves a voice message to John. With this option set to 'Yes', voice message will be deleted after sending it as an attachment to John's email address
Field Type: Option buttons

Say CallerID:

Should extension number which left the voice message be announced to mailbox owner

Example: With this option set to 'Yes', John will hear '... from phone number 1004...' when checking mailbox, for example.
Field Type: Option buttons

Allow Review mode:

Allow user to review his voice message before committing it permanently to voice mailbox

Example: After caller leaves the voice message and presses '#', additional review options are allowed: 1 to accept the recording, 2 to re-record your message etc...
Field Type: Option buttons

Allow Operator:

Allow caller to reach the operator from the voice inbox by pressing '0'

Example: Once user leave a voice message and presses #, additional options, including '...press 0 to reach an Operator' are heard
Field Type: Option buttons

Operator Extension:

Local extension number that is dialed once '0' is pressed to reach the Operator

Example: Once the caller leave a voice message to John and presses '0' to reach the Operator, extension number provided here (e.g. 1001) will be dialed
Field Type: [0-9]

Play Envelope Message:

Announces date and time when the voice message was left in inbox

Example: With this option enabled, John will hear 'First message, 11:52, 02 Feb 2007' for example, when checking his voice mailbox
Field Type: [0-9]

Voicemail Delay:

Delay time in seconds inserted before the Busy/Unavailable message is played to caller. This solves the 'half-played' file problem. Keep this value between 1-3

Example: Caller is to leave a voice message to John. It hears '...ot at home right now...'. Adding '1' to this field will add one second pause before the message is played. So, now new callers will hear the greeting message without the first part being cut off 'I am not at home right now...'.
Field Type: [0-9]

Timezone:

Set the correct date and time format for message envelope. NOTE: Timezones are taken from '/usr/share/zoneinfo' system directory.

Example: Some countries prefer time format in mm-dd-yy or dd-mm-yy format. Select among the available options
Field Type: Select box

Disk Space Used By Voicemail Recording:

With continuously tone 60 seconds:

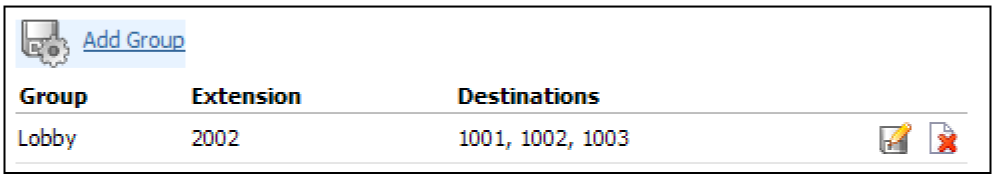
- wav49 = 91.0kb
- wav = 863.0kb
- gsm = 91.0kb

With continuously silent tone (without sound) 60 sec:

- wav49 = 0.38kb
- wav = 3.0kb
- gsm. = 0.32k b

11.3 Groups

Voicemail groups are used to group voicemail inboxes. Once a voice message is left to group 2002 for example, all destinations that belong to that voicemail group will receive the same voice message. This screen lists all system voicemail groups with the following details:



Group:
Voicemail group name/identifier
Example: Lobby
Field Type: Display

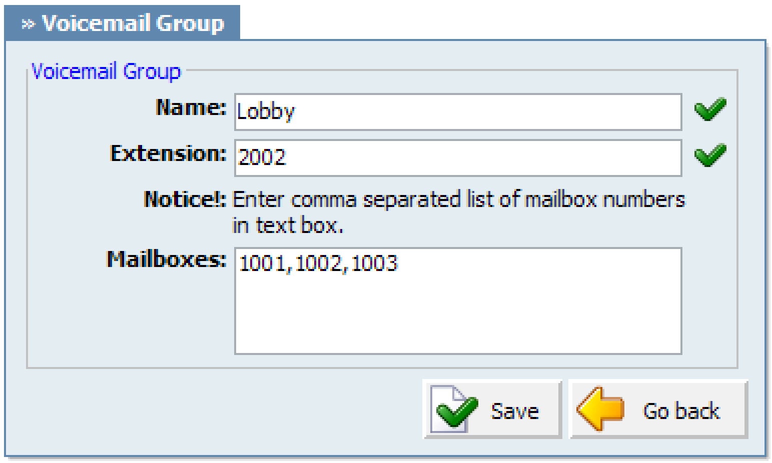
Extension:
Voicemail group extension. Once dialed, voicemail message will be left to all 'Destinations' voice boxes
Example: 2002
Field Type: Display

Destinations:
Voice boxes assigned to a voicemail group
Example: 1001, 1002, 1003
Field Type: Display

Edits the voicemail group configuration
Example: Click to edit voicemail group configuration
Field Type: Button

Deletes a voicemail group from the system
Example: Click to delete a voicemail group from the system
Field Type: Button

11.3.1 Add/Edit Group



Name:

Voice mailbox network name

Example: Create a voicemail group 'Lobby' for example
Field Type: [a-z][0-9]

Extension:

Voice mailbox network extension number

Example: To leave a message to this group mailbox callers will have to dial the extension number set here, e.g. 2002
Field Type: [0-9]

Mailboxes:

A list of mailboxes that belong to this voicemail group is set here

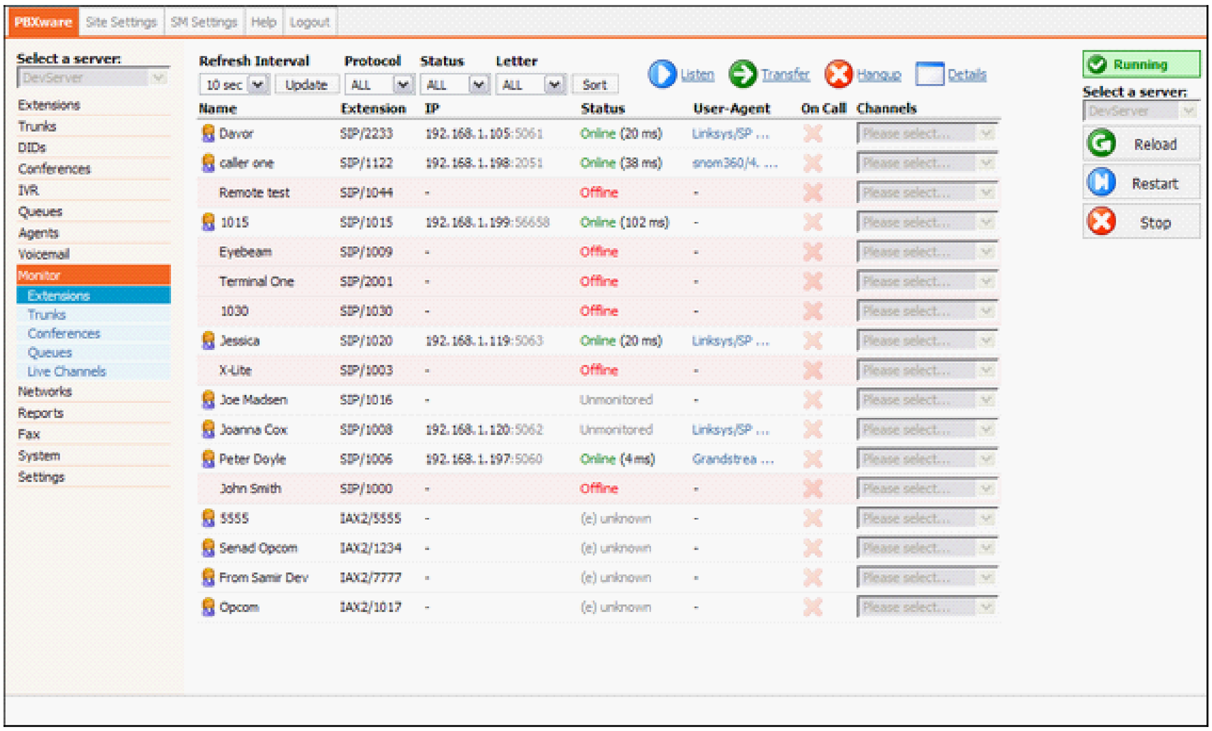
Example: To assign extensions to this voicemail group, provide their extension numbers separated by comma ',' here. For example 1001,1002,1003...
Field Type: [0-9]

Part XII

12 Monitor

Monitoring window allows administrator to monitor all PBXware extensions, trunks, conferences, queues and live channels in real time

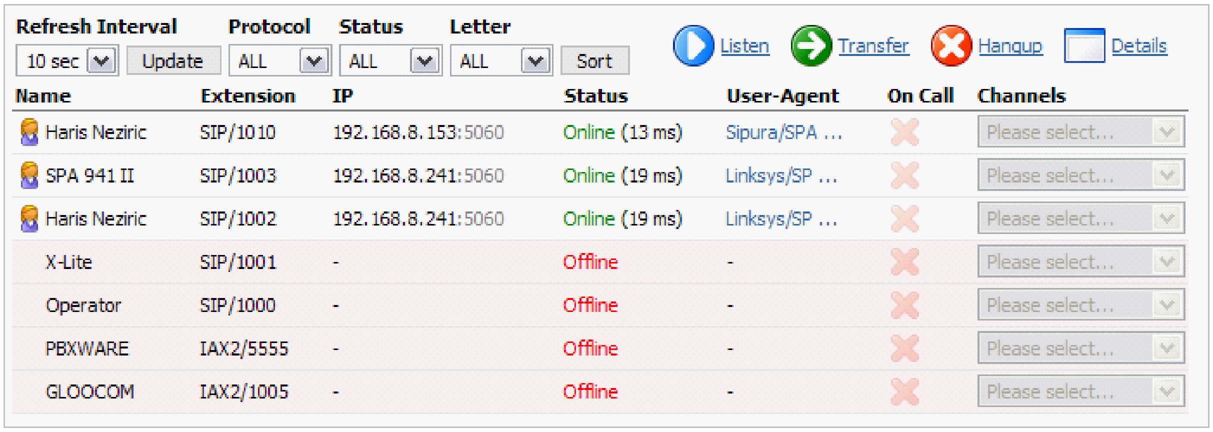
In this chapter we will cover:



12.1 Extensions

Monitored extensions are displayed in realtime with the following details:

NOTE: Certain call actions (such as transferring calls, hanging up etc...) can be performed on active calls as well



Name:
Name of the user extension is registered to
Example: Peter Doyle
Field Type: Display

Extension:
Protocol used by the extension/Extension network number
Example: SIP/2002
Field Type: Display

IP:
IP address:port UAD/Phone registers from
Example: 192.168.1.1:5060
Field Type: Display

Status:
UAD/Phone network status (Online/Offline) + (ping time)
Example: Online (56ms)/Offline
Field Type: Display

User Agent:
UAD/Phone Brand/Version
Example: Grandstream 101
Field Type: Display

On Call:
Is user participating in conversation at this moment
Example: -Yes, -No
Field Type: Display

Channels:
Shows the current channels available. Used with 'Listen', 'Transfer', 'Hangup' and 'Details' commands
Example: SIP/1111-1de6 (Bridged Call)
Field Type: Display

12.1.1 Search

Refresh Interval:

Time interval in seconds at which data details should be refreshed

Example: Select '10 sec' in this field for example and click the 'Update' button
Field Type: Select box

Protocol:

Filter the data based on the protocol type (ALL, SIP, IAX)

Example: Select 'ALL' for example, and click the 'Sort' button to display both SIP and IAX extensions
Field Type: Select box

Status:

Sort extensions based on their network status (ALL, Online, Offline)

Example: Select 'Online' for example, and click the 'Sort' button to display extensions that are registered/ online only
Field Type: Select box

Letter:

Sort extensions based on the user name they belong to

Example: Select 'B' for example, and click the 'Sort' button to display extensions that belong to users whose names start with letter B (e.g. Brown James)
Field Type: Select box

12.1.2 Actions

Listen:

Listen active conversations. Select one of the active conversations under 'Channels', click this button and provide extension number that is to listen active conversation.

NOTE: You may listen active conversations by dialing *199 + $EXTENSION number as well. But, no matter which method you listen the calls with, the listen service has to be enabled in the enhanced services of the extension that listens the call.

Example: Let's say that extensions 1000 and 1001 are in conversation. Select this button and type 1005 into popup window. Extension 1005 will ring and once the handset is picked up, active conversation will be heard.
Field Type: Button

Transfer:

Transfer a party from the active conversation to different destination

Example: Let's say that extensions 1000 and 1001 are in conversation. Select one extension (e.g. 1000) under 'Channels' and click this button. Type 1005 into popup window. Extension '1001' will be transferred to extension '1005'
Field Type: Button

Hangup:

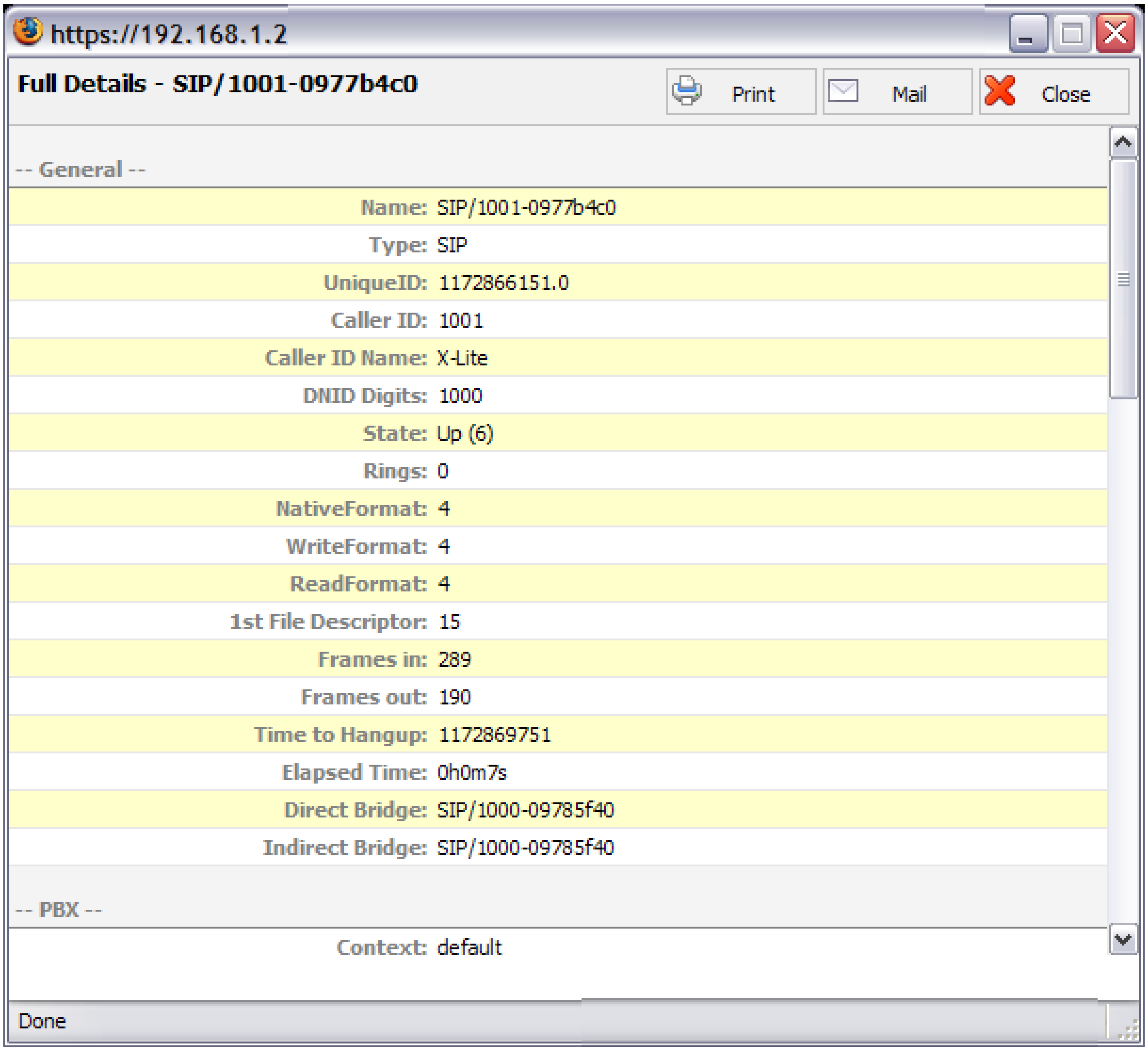
Hangup active conversation

Example: Let's say that extensions 1000 and 1001 are in conversation. Select one extension (e.g. 1000) under 'Channels' and click this button. Conversation between these two extensions will be terminated.
Field Type: Button

Details:

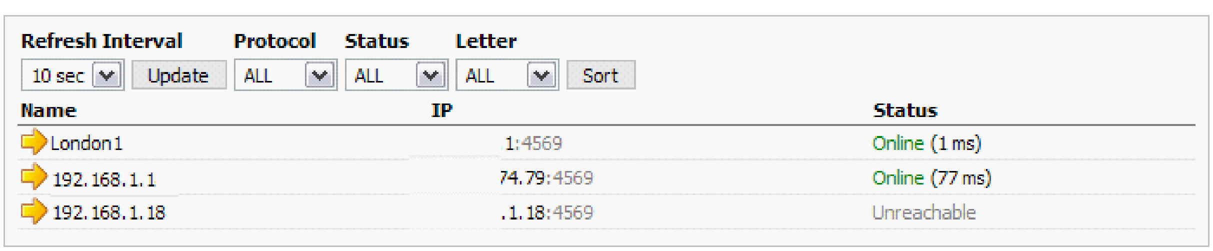
Display more technical details about the active call

Example: Let's say that extensions 1000 and 1001 are in conversation. Select one extension (e.g. 1000) under 'Channels' and click this button. New popup window will open with more details about the ongoing call
Field Type: Button



12.2 Trunks

Monitored trunks are displayed in realtime with the following details:



Name:

Trunk name

Example: Depending on the provider settings this can be set to a phone number, ip address or some context
Field Type: Display

IP:

Provider IP address

Example: 203.196.128.5
Field Type: Display

Status:

Displays the trunk status (online/offline) NOTE: Please set the 'Qualify' = '2500' in the Trunk settings to see its status

Example: If the 'Qualify' trunk option is empty, 'Unmonitored' is displayed here. Otherwise, '(e) ok (159ms)' is displayed, for example
Field Type: Display

12.2.1 Search

Refresh Interval:

Time interval in seconds at which data details should be refreshed

Example: Select '10 sec' in this field for example and click the 'Update' button
Field Type: Select box

Protocol:

Filter the data based on the protocol type (ALL, SIP, IAX)

Example: Select 'ALL' for example, and click the 'Sort' button to display both SIP and IAX extensions
Field Type: Select box

Status:

Sort extensions based on their network status (ALL, Online, Offline)

Example: Select 'Online' for example, and click the 'Sort' button to display extensions that are registered/ online only
Field Type: Select box

Letter:

Sort extensions based on the user name they belong to

Example: Select 'B' for example, and click the 'Sort' button to display extensions that belong to users whose names start with letter B (e.g. Brown James)
Field Type: Select box

12.3 Conferences

Monitored conferences are displayed in realtime with the following details:

Conferences

Privilege: Command

Conf Num	Parties	Marked	Activity	Creation
2255	0001	N/A	00:00:22	Static

* Total number of MeetMe users: 1

Conf Num:
Conference number
Example: 2255
Field Type: Display

Parties:
Number of participants in a conference
Example: 1001
Field Type: Display

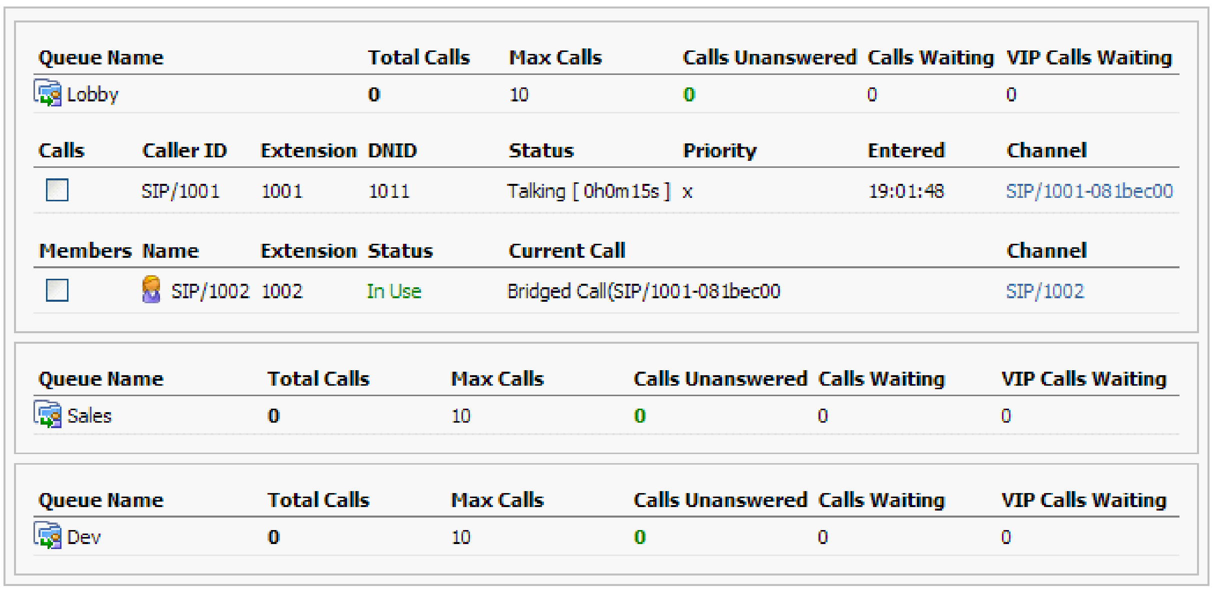
Marked:
Example:
Field Type:

Activity:
Time conference is active
Example: 00:00:22
Field Type: Display

Creation:
Conferences can be created dynamically and statically. Statically is pre-configured conference. Dynamic ones are created once the caller calls in, and are not currently supported
Example: Static/Dynamic
Field Type: Display

12.4 Queues

Monitored queues are displayed in realtime with the following details:



Queue Name	Total Calls	Max Calls	Calls Unanswered	Calls Waiting	VIP Calls Waiting
Lobby	0	10	0	0	0

Calls	Caller ID	Extension	DNID	Status	Priority	Entered	Channel
	SIP/1001	1001	1011	Talking [0h0m15s]	x	19:01:48	SIP/1001-081bec00

Members	Name	Extension	Status	Current Call	Channel
	SIP/1002	1002	In Use	Bridged Call(SIP/1001-081bec00	SIP/1002

Queue Name	Total Calls	Max Calls	Calls Unanswered	Calls Waiting	VIP Calls Waiting
Sales	0	10	0	0	0

Queue Name	Total Calls	Max Calls	Calls Unanswered	Calls Waiting	VIP Calls Waiting
Dev	0	10	0	0	0

Queue Name:

Queue name

Example: MainTest
Field Type: Display

Total Calls:

Total number of queue calls

Example: 0
Field Type: Display

Max Calls:

Maximum number of queue calls at the same time

Example: 4
Field Type: Display

Calls Unanswered:

Number of unanswered queue calls

Example: 0
Field Type: Display

Calls Waiting:

Number of the calls waiting in the queue

Example: 1
Field Type: Display

VIP Calls Waiting:

Number of VIP calls waiting in the queue

Example: 1
Field Type: Display

Calls:

Select box used with 'Action243' buttons

Example: Select this box and click on the action button
Field Type: Display

Caller ID:

Caller ID of user waiting in a queue (displayed in $TECHNOLOGY/$EXTENSION format)

Example: SIP/1020
Field Type: Display

Extension:

Extension number of the user waiting in a queue

Example: 1020
Field Type: Display

DNID:

Network extension number of the queue caller is waiting in

Example: 4000
Field Type: Display

Status:

CallerqQueue status displayed in '$Status [$Time]' format

Example: If caller is waiting in a queue 'Waiting [0h1m18s]' is displayed, and if caller is talking to a queue agent ' Talking[0h1m18s]' is displayed here
Field Type: Display

Entered:

Time user entered the queue in 'hh:mm:yy' format

Example: 14:59:40
Field Type: Display

Channel:

Network channel through which user connects to queue displayed in '$TECHNOLOGY/$EXTENSION-$UNIQUEID' format

Example: SIP/1020-09e8f7c8
Field Type: Display

Members:

Select box used with 'Action 243' buttons

Example: Select this box and click on one of the action button
Field Type: Display

Name:

Name of the queue member displayed in '$STATUS/$EXTENSION NUMBER' format

Example: Agent/4010
Field Type: Display

Extension:

Extension number queue member is connection from

Example: 1001
Field Type: Display

Status:

Status of the queue member

Example: If the queue member is talking, 'Busy' is displayed here
Field Type: Display

Current Call:

Status of the current queue member call

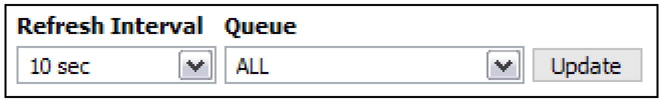
Example: If queue member is talking with extension 1020 for example, 'Bridged Call(SIP/1020-09e96618' is displayed here
Field Type: Display

Channel:

Network channel queue agent connects to the queue

Example: Agent/4010
Field Type: Display

12.4.1 Refresh



Refresh Interval:

Time interval in seconds at which data details should be refreshed

Example: Select '10 sec' in this field for example and click the 'Update' button
Field Type: Select box

Queue:

Select which queue data is to be displayed

Example: Select 'ALL' to display information on all PBXware queues, or select the queue name to view information about that queue only
Field Type: Select box

12.4.2 Actions

Listen:

Listen active conversations. Select one of the active conversations under 'Calls', click this button and provide extension number that is to listen active conversation.

NOTE: You may listen active conversations by dialing *199 + $EXTENSION number as well. But, no matter which method you listen the calls with, the listen service has to be enabled in the enhanced services of the extension that listens the call.

Example: Let's say that extension 1000 is waiting in queue. Select the box under 'Calls', click this button type 1005 into popup window. Extension 1005 will ring and once the handset is picked up, the conversation between ext 1000 and and queue member will be heard.
Field Type: Button

Transfer:

Transfer a party from the active conversation to different destination

Example: Let's say that extension 1000 is waiting in queue. Select the box under 'Calls', click this button and type 1005 into popup window. Extension 1000 will be transferred to extension 1005
Field Type: Button

Hangup:

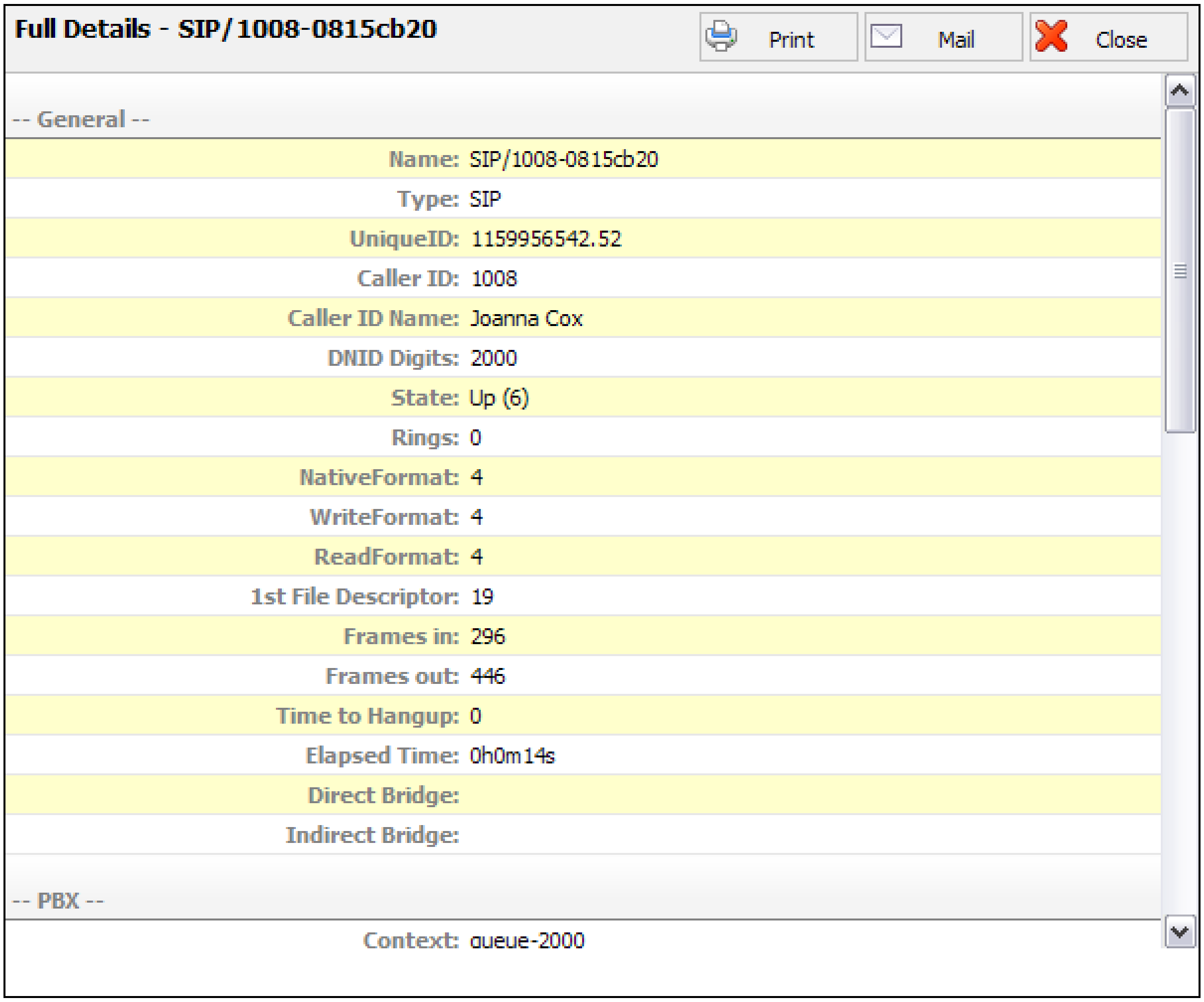
Hangup active conversation

Example: Let's say that extensions 1000 is talking with queue agent 1001. Select the extension 1000 under 'Calls' and click this button. Conversation between these two extensions will be terminated.
Field Type: Button

Details:

Display more technical details about the active call

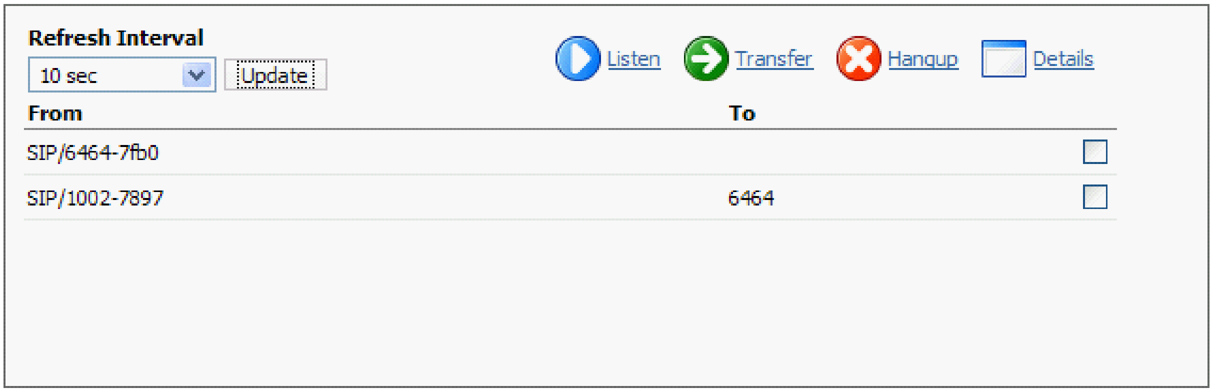
Example: Let's say that extensions 1000 and 1001 are in conversation. Select one extension (e.g. 1000) under 'Channels' and click this button. New popup window will open with more details about the ongoing call
Field Type: Button



12.5 Live Channels

Monitored live channels are displayed in realtime with the following details:

NOTE: Certain call actions (such as transferring calls, hanging up etc...) can be performed on active calls as well



From:

SIP/EXTENSION number of user making the call

Example: SIP/6464
Field Type: Display

To:

Extension number of user receiving a call

Example: SIP/6464
Field Type: Display

12.5.1 Refresh

Refresh Interval:

Time interval in seconds at which data details should be refreshed

Example: Select '10 sec' in this field for example and click the 'Update' button
Field Type: Select box

12.5.2 Actions

Listen:

Listen active conversations. Select the box next to one of the active conversations, click this button and provide Extension number that is to listen active conversation.

NOTE: You may listen active conversations by dialing *199 + $EXTENSION number as well. But, no matter which method you listen the calls with, the Listen service has to be enabled in the enhanced services of the extension that listens the call.

Example: Let's say that extension 1000 is waiting in queue. Select the box under 'Calls', click this button type 1005 into popup window. Extension 1005 will ring and once the handset is picked up, the conversation between ext 1000 and and queue member will be heard.
Field Type: Button

Transfer:

Transfer a party from the active conversation to different destination

Example: Let's say that extension 1000 is waiting in queue. Select the box under active call line, click this button and type 1005 into popup window. Extension 1000 will be transferred to extension 1005
Field Type: Button

Hangup:

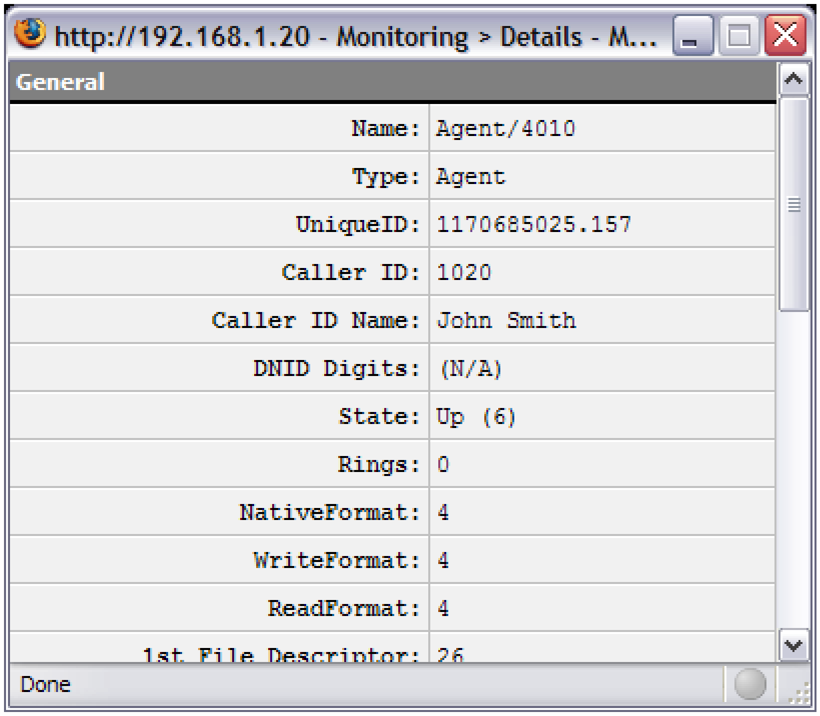
Hangup active conversation

Example: Let's say that extensions 1000 is talking with queue agent 1001. Select the box under the 1000 and click this button. Conversation between these two extensions will be terminated.
Field Type: Button

Details:

Display more technical details about the active call

Example: Let's say that extensions 1000 and 1001 are in conversation. Select the box under the 1000 and click this button. New popup window will open with more details about the ongoing call
Field Type: Button

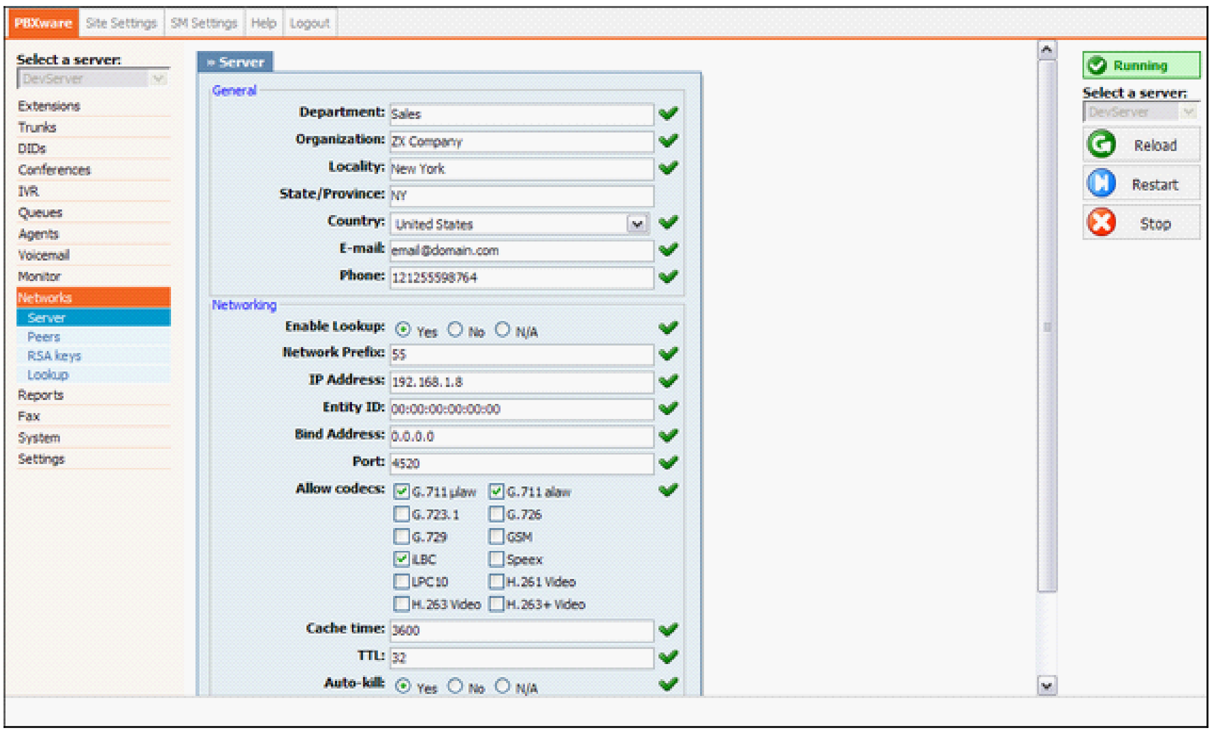

http://192.168.1.20 - Monitoring > Details - M...
General
Name: Agent/4010
Type: Agent
UniqueID: 1170685025.157
Caller ID: 1020
Caller ID Name: John Smith
DNID Digits: (N/A)
State: Up (6)
Rings: 0
NativeFormat: 4
WriteFormat: 4
ReadFormat: 4
1st File Descriptor: 26
Done

Part XIII

13 Networks

Networks are a transmission lines between two systems. But, unlike the trunks which use PSTN and VOIP technologies, networks use DUNDi(Distributed Universal Number Discovery) or something like a peer-to-peer protocol for communication with other systems over Internet gateways.

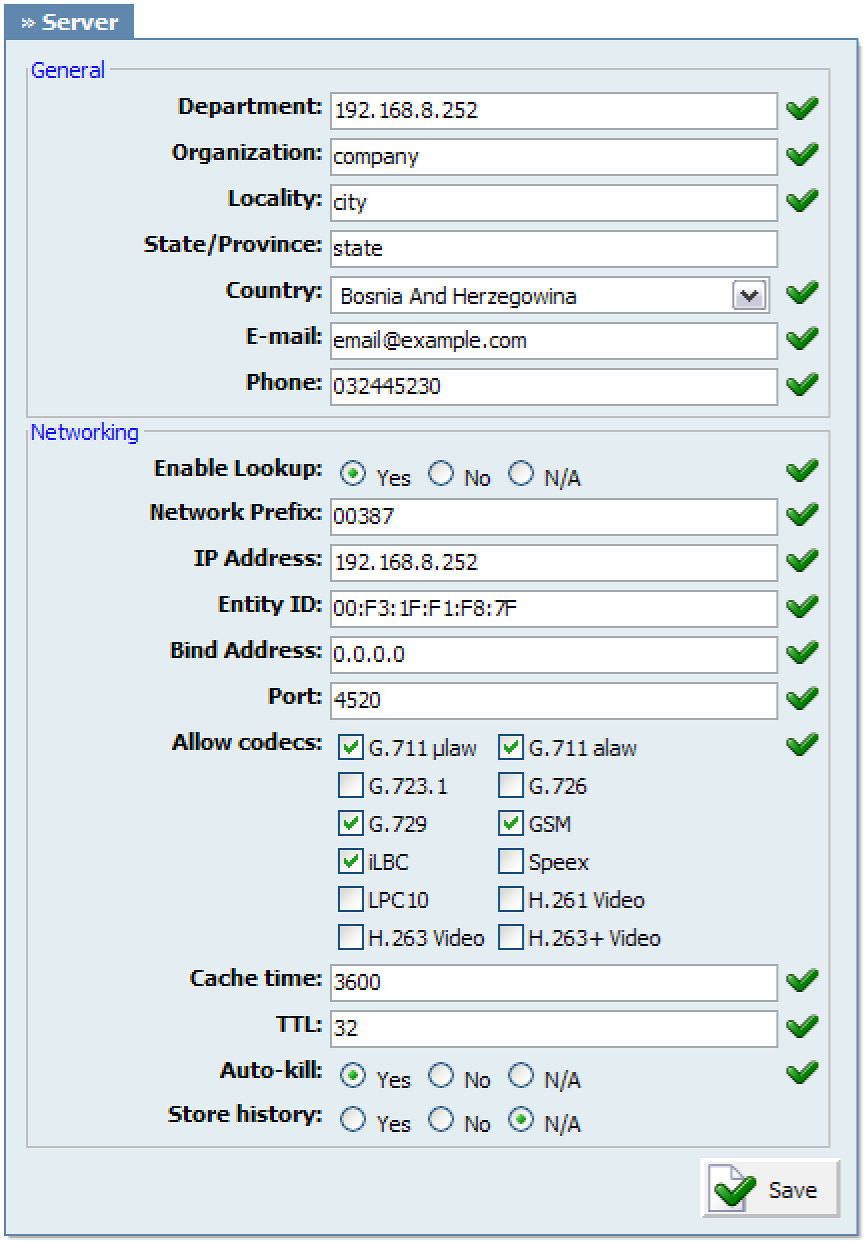
In this chapter we will cover:
- Server 250
- Peers 256
- RSA keys 258
- Lookup 259



13.1 Server

This window displays location and PBXware network information used by peers to connect to our system

In this chapter we will cover:
- General 251
- Networking 253



13.1.1 General

This section contains the PBXware location information.

Department:

Department name

Example: Sales Department
Field Type: [a-z][0-9]

Organization:

Organization name

Example: ZX Company
Field Type: [a-z][0-9]

Locality:

Company Surrounding/Nearby location

Example: NYC
Field Type: [a-z][0-9]

State/Province:

Company State/Province location

Example: NY
Field Type: [a-z][0-9]

Country:

Company Country location

Example: United States
Field Type: [a-z][0-9]

Email:

Company contact email address

Example: info@domain.com
Field Type: [a-z][0-9]

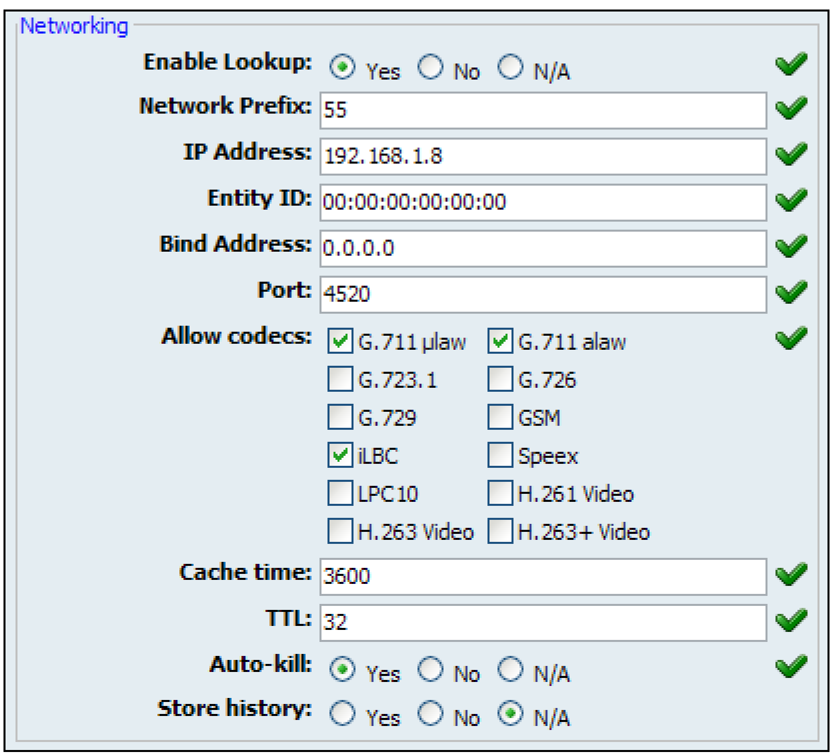
Phone:

Company contact phone number

Example: 2122443040
Field Type:[0-9]

13.1.2 Networking

This section contains the PBXware network details



Enable lookup:

Enable peer lookup for this server

Example: Setting this option to 'Yes' will allow other peers to find information about this server. It is recommended to keep this option set to 'Yes'
Field Type: Option buttons

Network Prefix:

Prefix number assigned to the server

Example: This number is similar to Area Code. It has to be dialed by other peers in order to access extensions on our side. Setting this option to '6' means that other peers will dial 6+1000 to dial extension 1000 on our server
Field Type: [0-9]

IP Address:

PBXware IP address other peers will connect to

Example: If PBXware is located on public IP address 89.223.12.93, type that IP here
Field Type: [0-9]

Entity ID:

PBXware MAC address. If PBXware has more then one MAC address, provide one of the first eth device

Example: 00:07:E9:3B:76:60
Field Type: [a-z][0-9]

Bind Address:

Set the Bind Address allowed to connect to our peer

Example: To allow all IP addresses to connect to us, set '0.0.0.0', or 192.168.1.20 to allow access from this IP address only
Field Type: [0-9]

Port:

Port through which other peers will connect to our server

Example: Default Bind Address Port is '4520'
Field Type: [0-9]

Codecs:

Codecs are allowed to other peers when connecting to our PBXware. See 'Codec Bandwidth' below for more

Example: It is recommended that you check the boxes next to ulaw, alaw, g719, gsm and ilbs codecs
Field Type: [0-9]

Cache:

Time in seconds during which peers will cache our query responses

Example: Default value '3600'
Field Type: [0-9]

TTL:

Time in milliseconds system will wait for the response

Example: Default value '32'
Field Type: [0-9]

Auto-kill:

Cancels the connection if there is no response within 2 seconds

Example: It is recommended that you keep this option set to 'Yes'
Field Type: Option buttons

Store history:

Should PBXware keep track of the last several queries and their execution time

Example: Set to 'Yes' only when debugging for it impacts the performance. Default value 'No'
Field Type: Option buttons

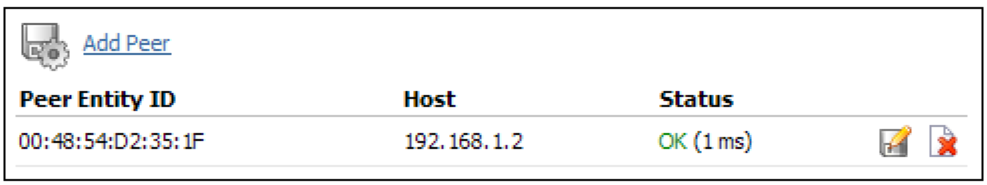
Codec Bandwidth:

- **ITU G.711 ulaw** **-** 64 Kbps, sample-based, used in US

- **ITU G.711 alaw** - 64 Kbps, sample-based, used in Europe
- **ITU G.723.1** - (5.3/6.3 Kbps, 30ms frame size
- **ITU G.726** - 16/24/32/40 Kbps
- **ITU G.729** - 8 Kbps, 10ms frame size
- **GSM** - 13 Kbps (full rate), 20ms frame size
- **iLBC** - 15Kbps,20ms frame size: 13.3 Kbps, 30ms frame size
- **Speex** - 2.15 to 44.2 Kbps
- **LPC10** - 2.5 Kbps
- **H.261 Video** - Used over ISDN lines with resolution of 352x288
- **H.263 Video** - Low-bit rate encoding solution for video conferencing
- **H.263+ Video** - Extension of H.263 that provides additional features that improve compression over packet switched networks.

13.2 Peers

This section contains information about peer systems PBXware connects to. This screen lists all system network peers with the following details



Peer Entity ID:
PBXware MAC address
Example: 00:07:E9:3B:76:60
Field Type: Display

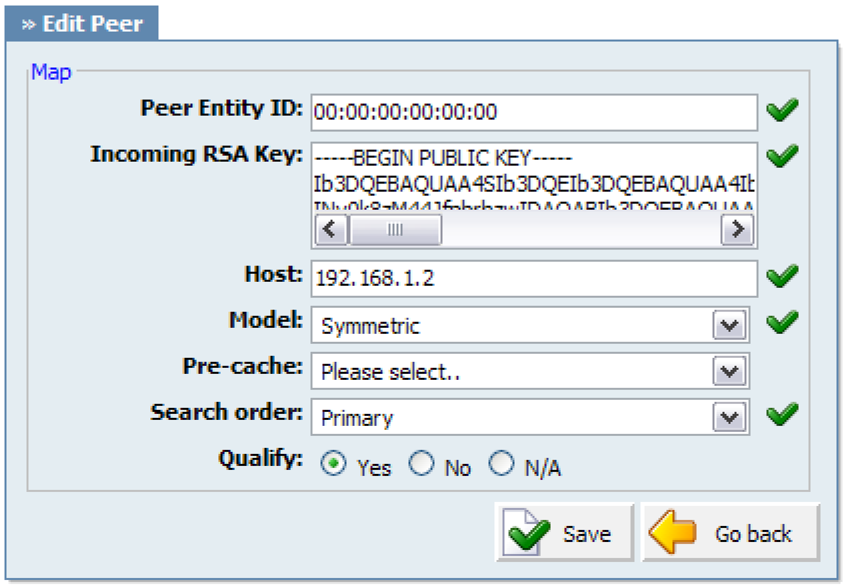
Host:
PBXware IP address
Example: 192.168.1.2
Field Type: Display

Status:
Peer connection status
Example: OK(1ms)
Field Type: Display


Edit peer settings
Example: Click to edit peer settings
Field Type: Button


Delete peer
Example: Click to delete peer from the system
Field Type: Button

13.2.1 Add/Edit



Peer Entity ID:

MAC address of the PBXware we are connecting to. If PBXware has more then one MAC address, provide one of the first eth device

Example: 00:07:E9:3B:76:60
Field Type: [a-z][0-9]

Incoming RSA Key:

RSA key of the PBXware we are connecting to

Example:
-----BEGIN PUBLIC KEY-----
MIGfMA0GCSqGSIbMIGfMA0GCSqGSIbMIGfMA0GCSqGSIbMIGfMA0GCSqGSIbMIGfMA
MIGfMA0GCSqGSIbMIGfMA0GCSqGSIbMIGfMA0GCSqGSIbMIGfMA0GCSqGSIbMIG
MIGfMA0GCSqGSIbMIGfMA0GCSqGSIbMIGfMA0GCSqGSIbMIGfMA0GCSqGSIb
MIGfMA0GCSqGSIbMIGfMA0GCS
-----END PUBLIC KEY-----
Field Type: [a-z][0-9]

Host:

Peer IP address PBXware is connecting to

Example: If PBXware is located on public IP address 89.223.12.93, type that IP here
Field Type: [0-9]

Pre-cache:

Set the peer pre-caching method

Example: Select among available options:

- **incoming** - permit peer to send pre-cache routes
- **outgoing** - send pre-cache routes to this peer
- **Symmetric** - both

Field Type: Select box

Search order:

Set the search order

Example: Select among available options:

- Primary
- Secondary
- Tertiary
- Quartiary

Field Type: Select box

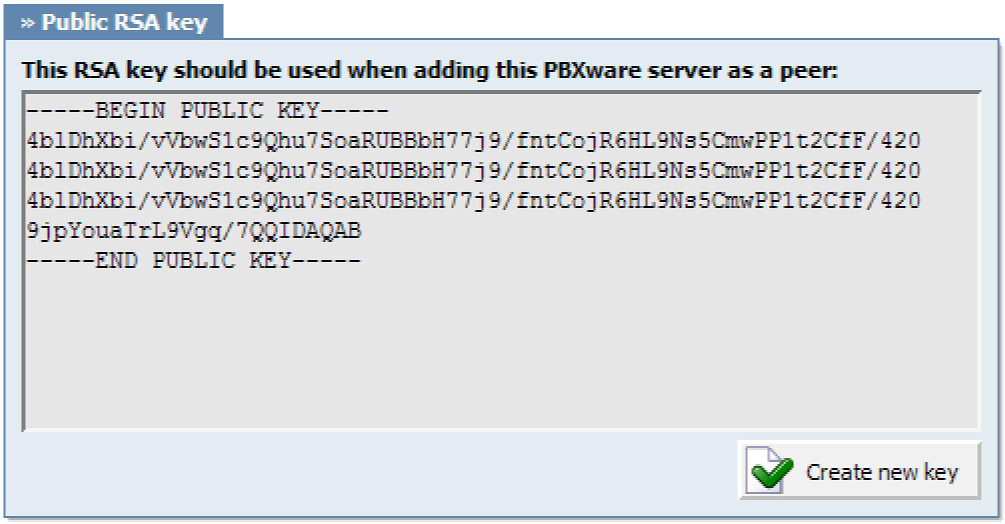
Qualify:

Should PBXware test if peer is alive

Example:
Field Type: Option buttons

13.3 RSA keys

This is the PBXware RSA key used for authentication with all network peers



Create new key

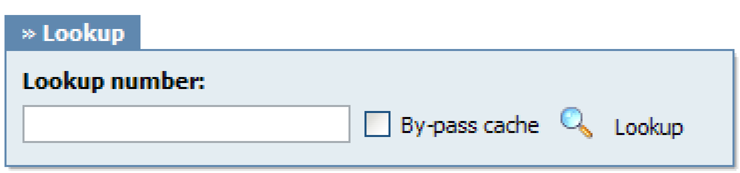
Generate new RSA key. This key will be used for PBXware connection authentication.

NOTE: Do not generate new RSA keys if connected to peers. If you do generate a new RSA key, be sure to pass it to all server you are registered with as peer

Example:
-----BEGIN PUBLIC KEY-----
MIGfMA0GCSqGSIbMIGfMA0GCSqGSIbMIGfMA0GCSqGSIbMIGfMA0GCSqGSIbMIGfMA
MIGfMA0GCSqGSIbMIGfMA0GCSqGSIbMIGfMA0GCSqGSIbMIGfMA0GCSqGSIbMIG
MIGfMA0GCSqGSIbMIGfMA0GCSqGSIbMIGfMA0GCSqGSIbMIGfMA0GCSqGSIb
MIGfMA0GCSqGSIbMIGfMA0GCS
-----END PUBLIC KEY-----
Field Type: [a-z][0-9]

13.4 Lookup

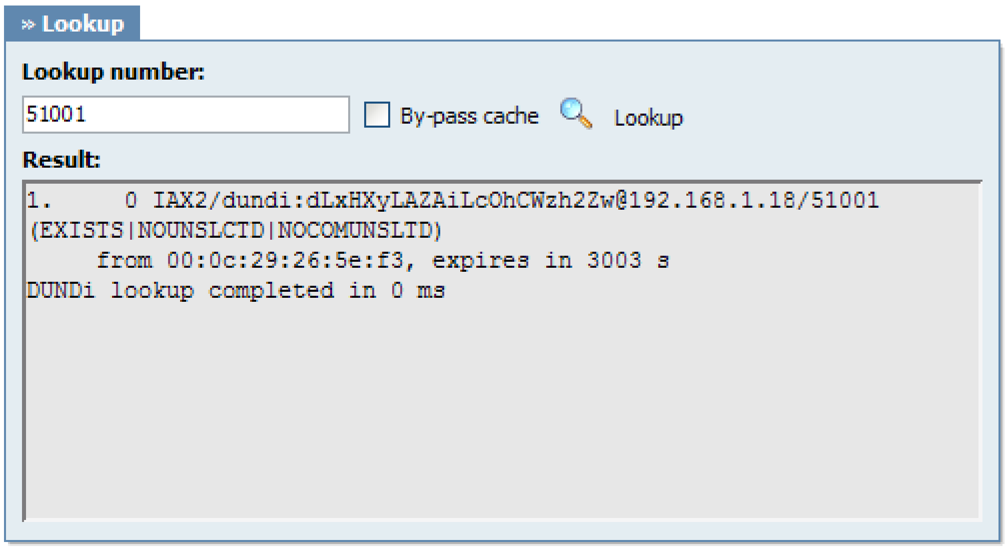
DUNDi lookup feature determines if a number can be reached via the service. Additional information is displayed commenting what is actually happening with the call



Lookup number:

Provide a fool lookup number here

Example: $NETWORKPREFIX + $NETWORK NUMBER (e.g. 51001. 5=Network prefix and 1001 is local extension)
Field Type: [0-9]



By-pass cache:

By-pass cache when performing lookup

Example: It is recommended to keep this option checked
Field Type: Option button

Lookup

Select this button to perform the lookup. The result will be displayed in new window below

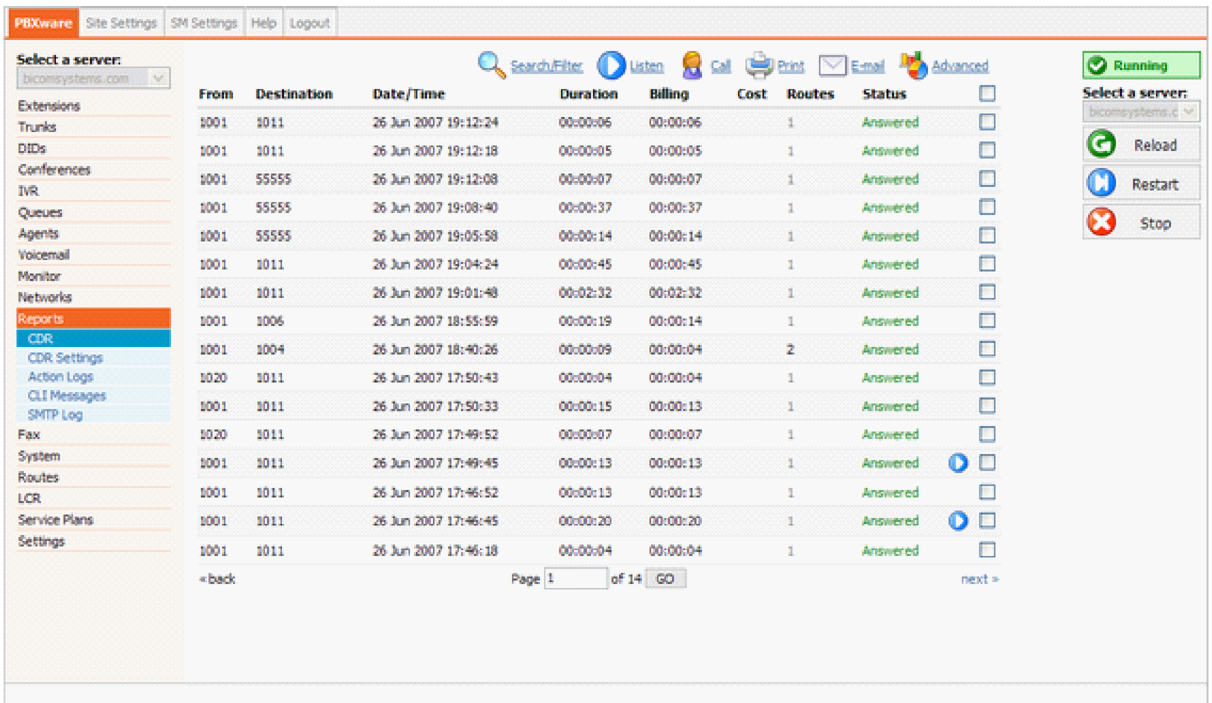
Example: Provide the lookup number, by-pass the cache and click this button to perform a lookup
Field Type: Button

Part XIV

14 Reports

Reports display detail records of all PBXware calls, system action logs, CLI messages and SMTP logs

In this chapter we will cover:
- Action Logs 270
- CLI Messages 272
- CDR Settings 270



14.1 CDR

CDR displays detail records of all PBXware calls with the following details

Search/Filter Listen Call Print E-mail Advanced

From	Destination	Date/Time	Duration	Billing	Cost	Routes	Status
1001	1011	26 Jun 2007 19:12:24	00:00:06	00:00:06		1	Answered
1001	1011	26 Jun 2007 19:12:18	00:00:05	00:00:05		1	Answered
1001	55555	26 Jun 2007 19:12:08	00:00:07	00:00:07		1	Answered
1001	55555	26 Jun 2007 19:08:40	00:00:37	00:00:37		1	Answered
1001	55555	26 Jun 2007 19:05:58	00:00:14	00:00:14		1	Answered
1001	1011	26 Jun 2007 19:04:24	00:00:45	00:00:45		1	Answered
1001	1011	26 Jun 2007 19:01:48	00:02:32	00:02:32		1	Answered
1001	1006	26 Jun 2007 18:55:59	00:00:19	00:00:14		1	Answered
1001	1004	26 Jun 2007 18:40:26	00:00:09	00:00:04		2	Answered
1020	1011	26 Jun 2007 17:50:43	00:00:04	00:00:04		1	Answered
1001	1011	26 Jun 2007 17:50:33	00:00:15	00:00:13		1	Answered
1020	1011	26 Jun 2007 17:49:52	00:00:07	00:00:07		1	Answered
1001	1011	26 Jun 2007 17:49:45	00:00:13	00:00:13		1	Answered
1001	1011	26 Jun 2007 17:46:52	00:00:13	00:00:13		1	Answered
1001	1011	26 Jun 2007 17:46:45	00:00:20	00:00:20		1	Answered
1001	1011	26 Jun 2007 17:46:18	00:00:04	00:00:04		1	Answered

« back Page 1 of 14 GO next »

From:
Extension number the call was made from
Example: If call was made from extension 1001 to extension 1004, '1001' is displayed here.
Field Type: Display

Destination:
Extension number the call was made to
Example: If call was made from extension 1001 to extension 1004, '1004' is displayed here.
Field Type: Display

Date/Time:
Date and Time when the call was made
Example: 04 Oct 2006 10:44:10
Field Type: Display

Duration:
Call duration time in hh:mm:ss format
Example: 00:12:45
Field Type: Display

Billing:
Time billed by the system
Example: 00:12:45
Field Type: Display

Routes:
Number of system routes
Example: If call goes directly from extension to destination - '1' will be displayed here. If call has entered

the IVR and was then redirected to queue and then agent answers the call '2' will be displayed here (IVR and queue were used)
Field Type: Display

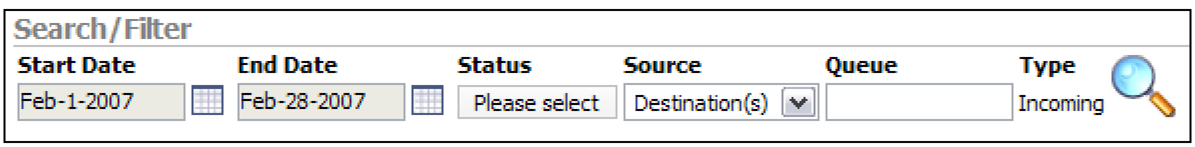
Status:
Displays the call status
Example: Depending on whether a call was answered or not, this field value may have the following content:
- Answered
- Not Answered
- Busy
- Error

Field Type: Display


This icon is displayed once a call is recorded and 'Delete' or 'Listen' enhanced service is active


This is a box used with the CDR commands to select a desired call

14.1.1 Search/Filter



Start Date:

Select a Search/Filter start date

Example: Click on a small 'Calendar' icon next to a field and select desired date
Field Type: Option button

End Date:

Select a Search/Filter end date

Example: Click on a small 'Calendar' icon next to a field and select desired date
Field Type: Option button

Status:

Search calls by selecting desired call status

Example: Click on a 'Please Select' button and select one of the available fields:
- All

- Answered
- Not Answered
- Busy
- Error

Field Type: Select box

Source:

Search calls by source type

Example: Select 'Destinations' to display calls that came over network extensions or 'Trunk' to display calls that came from other providers/PBXwares etc...
Field Type: Option button

Type:

Search calls based on the type of calls

Example: Click the 'Type' button and select one of the available fields:
- All
- Outgoing
- Incoming

Field Type: Select box

NOTE: After making any changes to search filter, be sure to click the  search icon

14.1.2 Actions

In this chapter we will cover:

14.1.2.1 Listen

Listen:

Once the 'Listen'  icon is displayed next to a call record it means that the specific call was recorded.

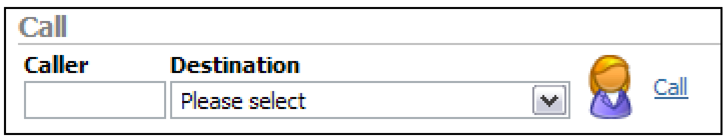
Example: To play recorded calls, check the box next to a 'Listen' icon and click 'Listen'. Browser will prompt you to open the sound file in your favorite audio player or to download the sound file.
Field Type: Option button

NOTE: By default the sound format is available as a .gsm file. To change the recording format go to: 'Settings: Servers: Edit: Recordings format' and select one of the available sound formats:

- gsm
- wav
- wav49 and
- ogg

14.1.2.2 Call

To establish a call between two PBXware extensions all you need to provide is the caller $EXTENSION number and the $DESTINATION extension



Caller

PBXware extension that will make a call

Example: Provide any PBXware extension number here, 1001 for example
Field Type: [0-9]

Destination:

Destination extension that will be dialed by 'Caller' extension

Example: To select a destination extension, first check a box next to a CDR record. This field will display two extensions listed under 'From' and 'Destination' selected record
Field Type: Select button

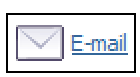
NOTE: After setting 'Caller' and 'Destination' extensions click the  call icon

14.1.2.3 Print

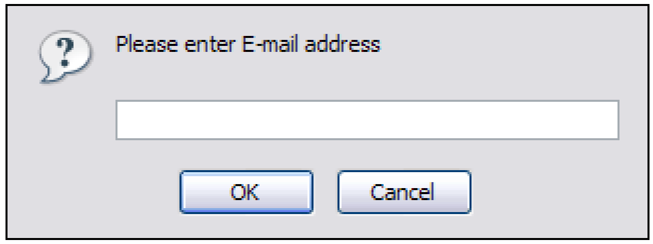
Check the box next to a call record and click the 'Print' button. This action will open a new popup window with the printing interface.

From	Destination	Date/Time	Duration	Status
5555	4000	30 Jan 2007 11:25:02	00:00:24	Answered
1000	4000	29 Jan 2007 17:21:20	00:00:14	Answered
1000	4000	29 Jan 2007 17:20:45	00:00:25	Answered
1000	4000	29 Jan 2007 17:03:13	00:00:27	Answered
1000	4000	29 Jan 2007 17:02:47	00:00:14	Answered
1000	4000	29 Jan 2007 16:59:52	00:00:24	Answered
1000	4000	29 Jan 2007 16:59:50	00:00:09	Answered
1000	4000	29 Jan 2007 16:59:49	00:00:00	Answered
1000	4000	29 Jan 2007 16:59:29	00:00:19	Answered
1000	4000	29 Jan 2007 16:59:20	00:00:27	Answered
1000	4000	29 Jan 2007 16:58:48	00:00:38	Answered
1000	4000	29 Jan 2007 16:58:45	00:00:33	Answered
1000	4000	29 Jan 2007 16:58:33	00:00:13	Answered
1000	4000	29 Jan 2007 16:58:22	00:00:21	Answered
1000	4000	29 Jan 2007 16:58:12	00:00:15	Answered
1000	4000	29 Jan 2007 16:57:50	00:00:19	Answered

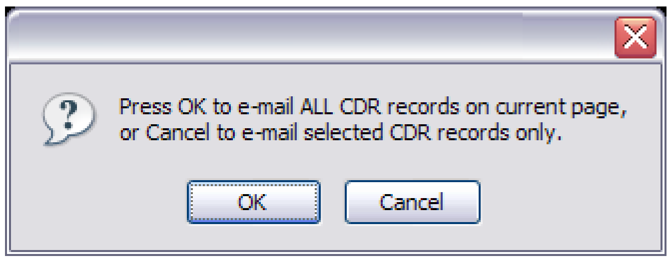
14.1.2.4 Email



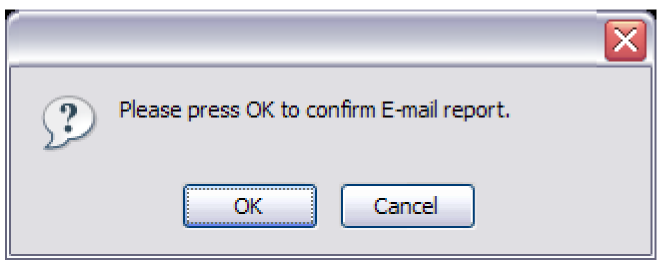
Click on 'Email' button to send all reports listed on page or
Select a box next to a report and click 'Email' button to send only selected ones



Provide E-mail address where report is to be sent and click 'OK' button to proceed or 'Cancel' to abort the email action



Press 'OK' to email all CDR records on the current page (even if they are not selected) or click 'Cancel' to print selected records only

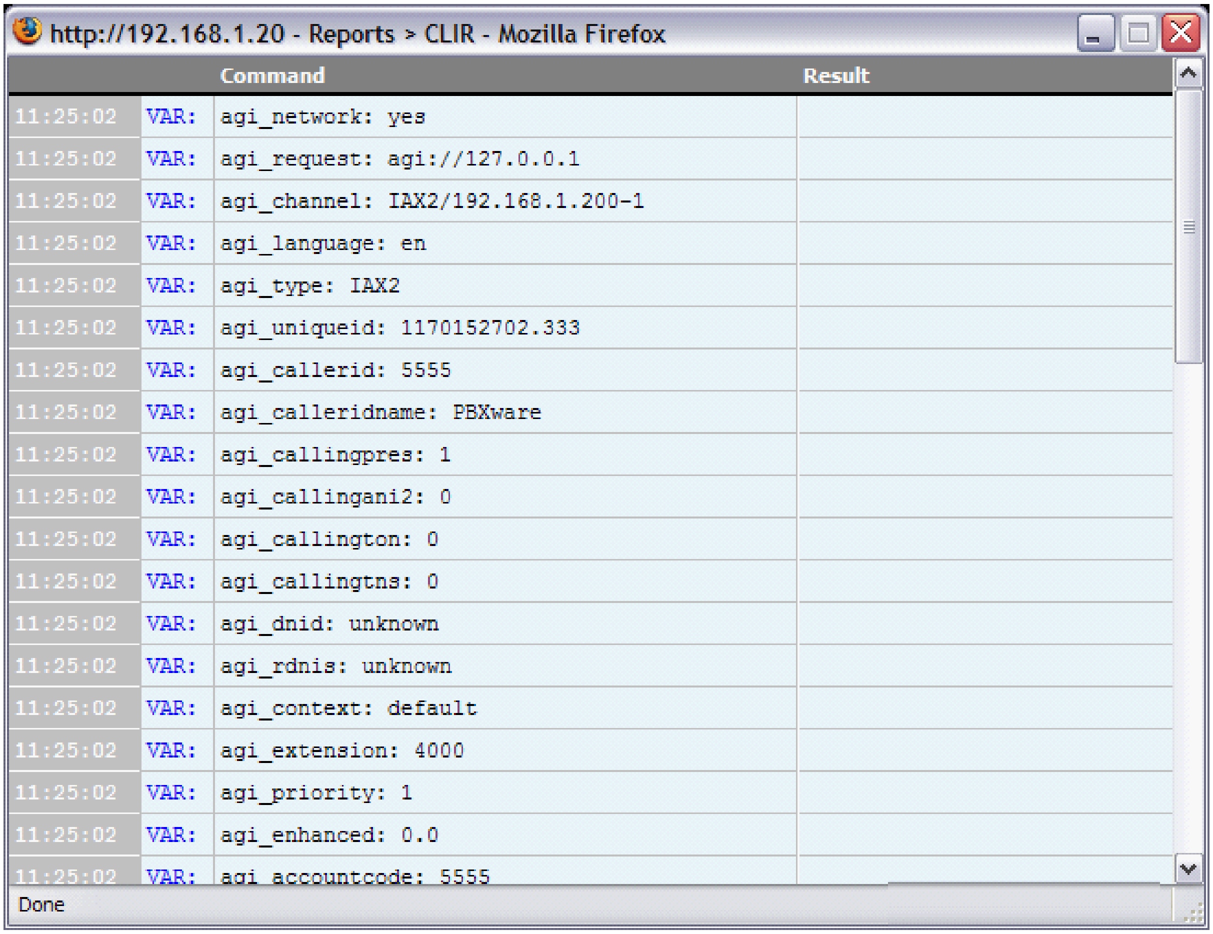


Finally, press 'OK' button to confirm email action or 'Cancel' to abort the email action

14.1.2.5 Advanced

In this chapter we will cover:
- CLIR 209
- Delete Recording 210
- Download CSV 210

14.1.2.5.1 CLIR



CLIR:

CLIR (Command Line Interface Record) details

Example: Select a desired call record and click this button to view more technical details about the call. A small popup window will open with the data. NOTE: When experiencing any kind of unexplained problems, this is the data you need to send to the technical support team
Field Type: Command Button

14.1.2.5.2 Delete Recording

Delete Recording:

Deletes the recorded calls. NOTE: For this command to be displayed, appropriate enhanced service has to be set.

Example: Select a recorded call and click this button to delete it from the file system
Field Type: Command Button

14.1.2.5.3 Download CSV

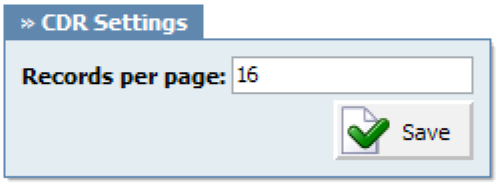
Download CSV:

Download data as the .csv (Comma Separated Value) file

Example: Click this button to download the .csv file to your desktop
Field Type: Command Button

14.2 CDR Settings

Please provide a number in 'Records per page' field



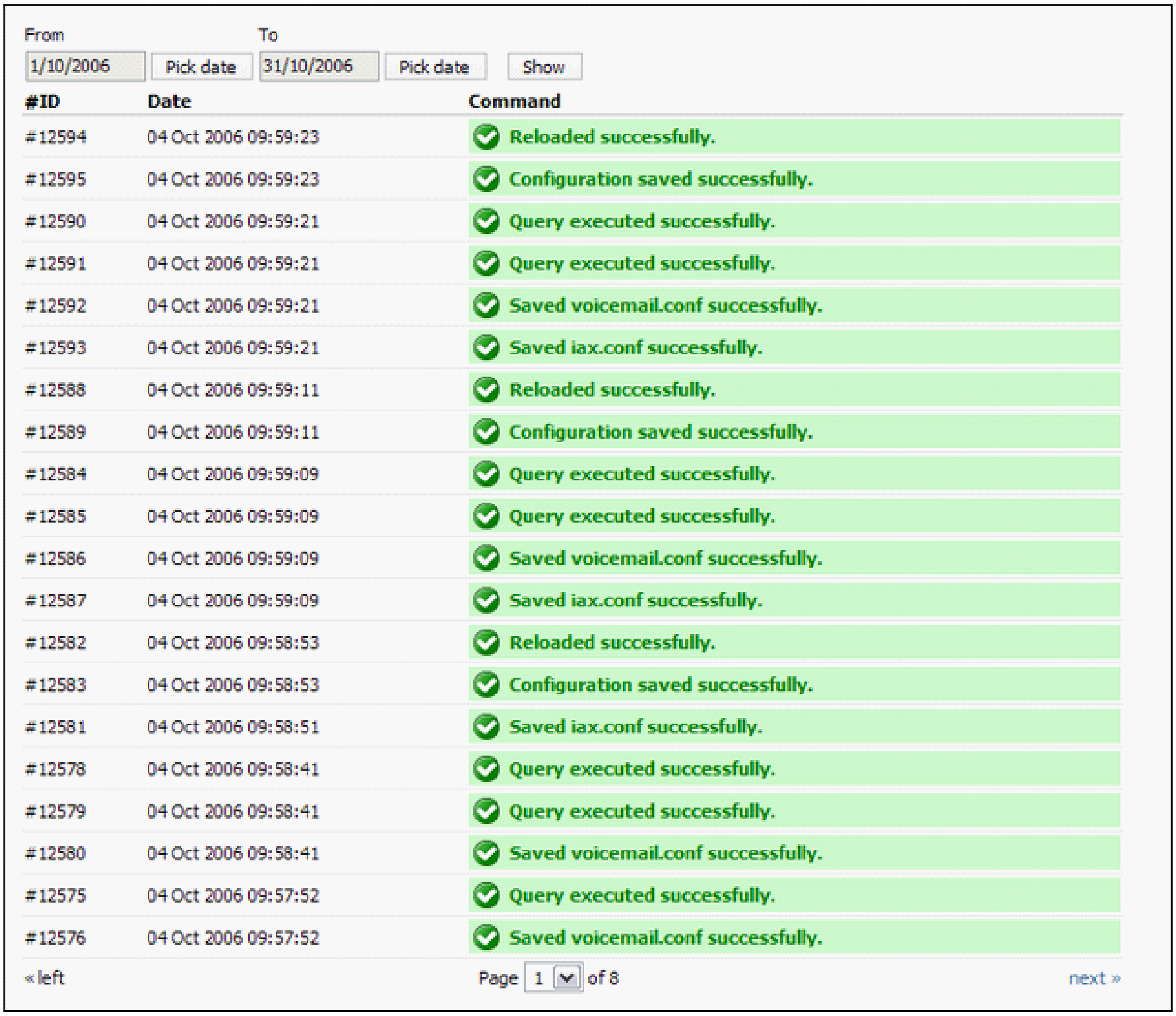
Records per page:

Number of records displayed per page

Example: When on 'Reports: CDR' page, and this option is set to '16', last 16 call records will be displayed. On the bottom there is 'Page' field. Type a page number, e.g. '2', and click 'GO' button to display next 16 call records
Field Type: [0-9]

14.3 Action Logs

Each time a user interacts with the system, its action is sent to an "action queue". The action queue then sends a command to the PBXware service that is running on either local or remote host(s). These commands are then analyzed by the action queue for its status. All successful commands are colored in green. All unsuccessful commands are colored in red, and queued for re-sending 5 minutes subsequently.



From 1/10/2006 Pick date To 31/10/2006 Pick date Show

#ID	Date	Command
#12594	04 Oct 2006 09:59:23	Reloaded successfully.
#12595	04 Oct 2006 09:59:23	Configuration saved successfully.
#12590	04 Oct 2006 09:59:21	Query executed successfully.
#12591	04 Oct 2006 09:59:21	Query executed successfully.
#12592	04 Oct 2006 09:59:21	Saved voicemail.conf successfully.
#12593	04 Oct 2006 09:59:21	Saved iax.conf successfully.
#12588	04 Oct 2006 09:59:11	Reloaded successfully.
#12589	04 Oct 2006 09:59:11	Configuration saved successfully.
#12584	04 Oct 2006 09:59:09	Query executed successfully.
#12585	04 Oct 2006 09:59:09	Query executed successfully.
#12586	04 Oct 2006 09:59:09	Saved voicemail.conf successfully.
#12587	04 Oct 2006 09:59:09	Saved iax.conf successfully.
#12582	04 Oct 2006 09:58:53	Reloaded successfully.
#12583	04 Oct 2006 09:58:53	Configuration saved successfully.
#12581	04 Oct 2006 09:58:51	Saved iax.conf successfully.
#12578	04 Oct 2006 09:58:41	Query executed successfully.
#12579	04 Oct 2006 09:58:41	Query executed successfully.
#12580	04 Oct 2006 09:58:41	Saved voicemail.conf successfully.
#12575	04 Oct 2006 09:57:52	Query executed successfully.
#12576	04 Oct 2006 09:57:52	Saved voicemail.conf successfully.

« left Page 1 of 8 next »

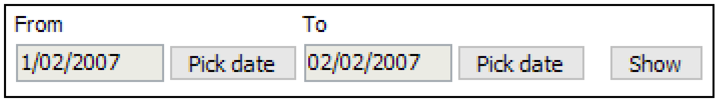
ID:
Action logs unique id (Identities)
Example: #4250
Field Type: Display

Date:
Date/Time action log has been performed
Example: 14-11-2005 15:12:23
Field Type: Button

Command:
Command sent to PBXware service

Example: Query executed successfully.
Field Type: Display

14.3.1 Search



From:

Select a Search/Filter start date

Example: Click on a 'Pick date' button next to a field and select desired date
Field Type: Option button

To:

Select a Search/Filter end date

Example: Click on a 'Pick date' button next to a field and select desired date
Field Type: Option button

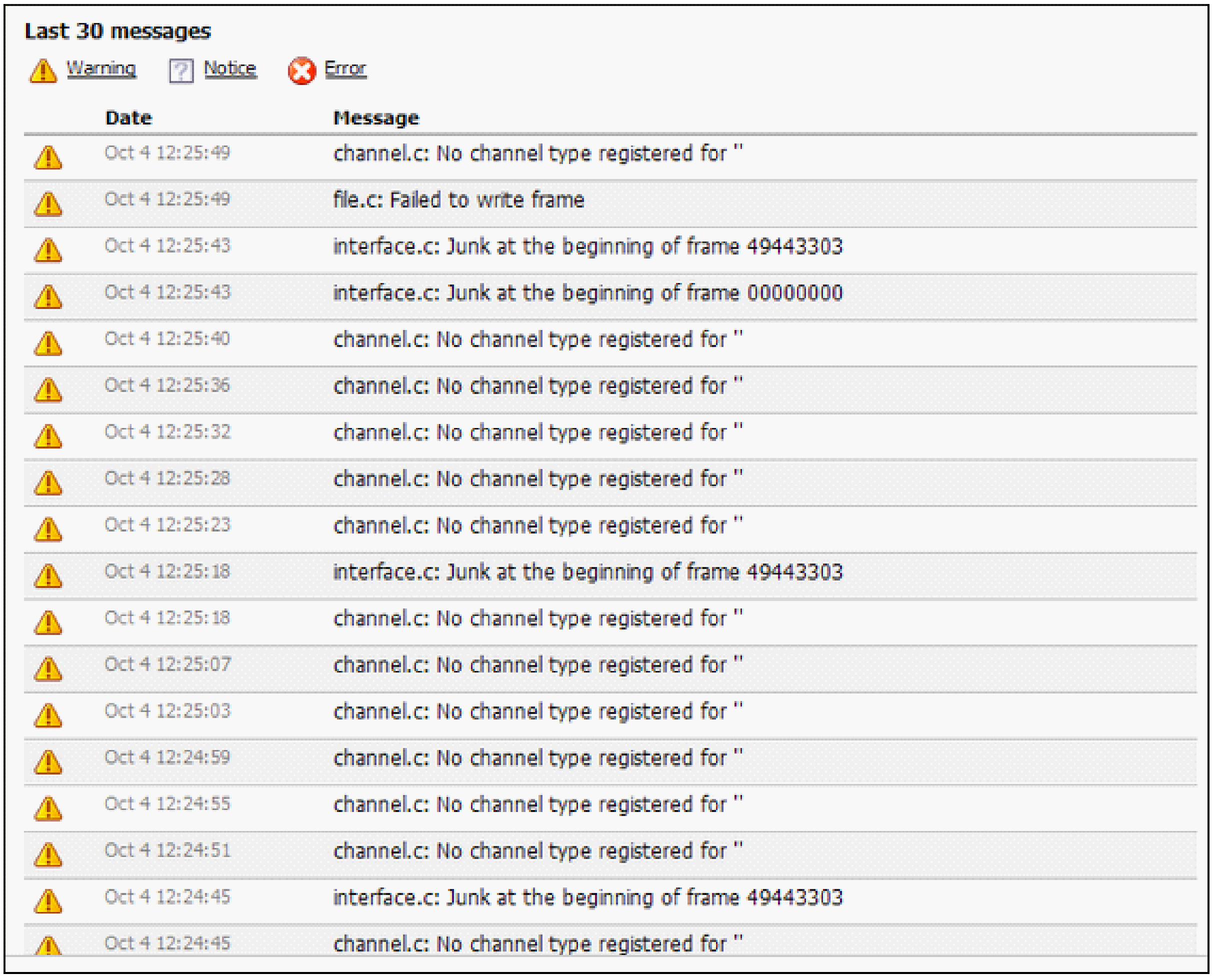
Show:

Display results based on the set Search details

Example: Select 'From' and 'To' date ranges and click this button to display the results
Field Type: Button

14.4 CLI Messages

CLI messages provide a convenient method of showing messages received from asterisk CLI (Command Line Interface). Each message is shown in the order received and if clicked on, it will open a new browser searching www.google.com with the message content text.



Last 30 messages

Warning Notice Error

	Date	Message
	Oct 4 12:25:49	channel.c: No channel type registered for ''
	Oct 4 12:25:49	file.c: Failed to write frame
	Oct 4 12:25:43	interface.c: Junk at the beginning of frame 49443303
	Oct 4 12:25:43	interface.c: Junk at the beginning of frame 00000000
	Oct 4 12:25:40	channel.c: No channel type registered for ''
	Oct 4 12:25:36	channel.c: No channel type registered for ''
	Oct 4 12:25:32	channel.c: No channel type registered for ''
	Oct 4 12:25:28	channel.c: No channel type registered for ''
	Oct 4 12:25:23	channel.c: No channel type registered for ''
	Oct 4 12:25:18	interface.c: Junk at the beginning of frame 49443303
	Oct 4 12:25:18	channel.c: No channel type registered for ''
	Oct 4 12:25:07	channel.c: No channel type registered for ''
	Oct 4 12:25:03	channel.c: No channel type registered for ''
	Oct 4 12:24:59	channel.c: No channel type registered for ''
	Oct 4 12:24:55	channel.c: No channel type registered for ''
	Oct 4 12:24:51	channel.c: No channel type registered for ''
	Oct 4 12:24:45	interface.c: Junk at the beginning of frame 49443303
	Oct 4 12:24:45	channel.c: No channel type registered for ''

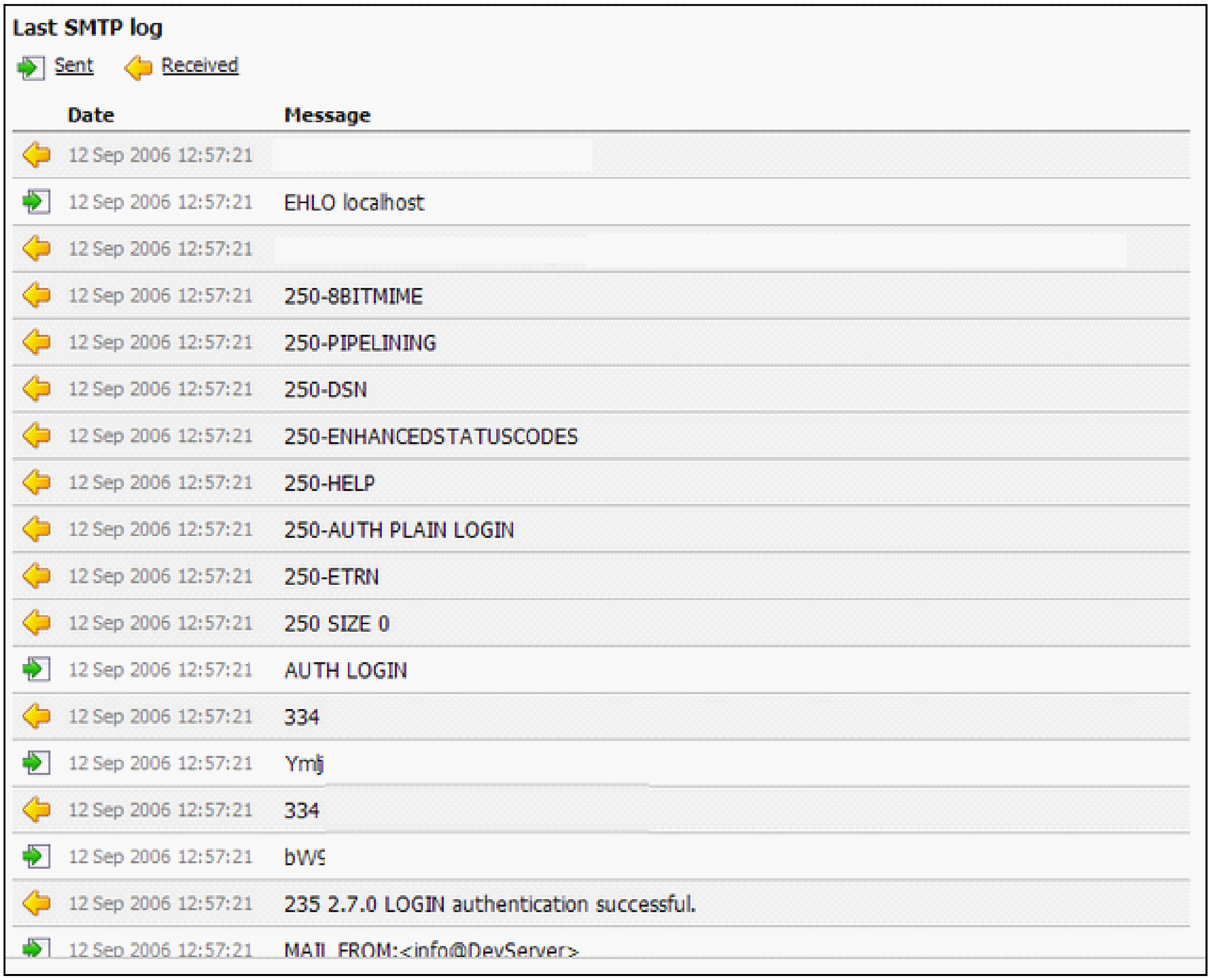
Available Message types:

- **Warning -** A warning message of an issue that will not usually affect the system's operation
- **Notice -** A notice message is simply a formal notice and does not affect the system's operation
- **Error -** Error message may in some situations stop or affect the system's operation

14.5 SMTP Log

Last messages archived in the SMTP log. Messages are marked as:

- Sent - Sent by PBXware and
- Received - Response from the SMTP server



Last SMTP log

Sent Received

	Date	Message
Received	12 Sep 2006 12:57:21	
Sent	12 Sep 2006 12:57:21	EHLO localhost
Received	12 Sep 2006 12:57:21	
Received	12 Sep 2006 12:57:21	250-8BITMIME
Received	12 Sep 2006 12:57:21	250-PIPELINING
Received	12 Sep 2006 12:57:21	250-DSN
Received	12 Sep 2006 12:57:21	250-ENHANCEDSTATUSCODES
Received	12 Sep 2006 12:57:21	250-HELP
Received	12 Sep 2006 12:57:21	250-AUTH PLAIN LOGIN
Received	12 Sep 2006 12:57:21	250-ETRN
Received	12 Sep 2006 12:57:21	250 SIZE 0
Sent	12 Sep 2006 12:57:21	AUTH LOGIN
Received	12 Sep 2006 12:57:21	334
Sent	12 Sep 2006 12:57:21	Ymlj
Received	12 Sep 2006 12:57:21	334
Sent	12 Sep 2006 12:57:21	bW9
Received	12 Sep 2006 12:57:21	235 2.7.0 LOGIN authentication successful.
Sent	12 Sep 2006 12:57:21	MAIL FROM:<info@DevServer>

Date:

Date/Time SMTP log was created

Example: 12 Sep 2006 12:57:21
Field Type: Display

Message:

SMTP server response

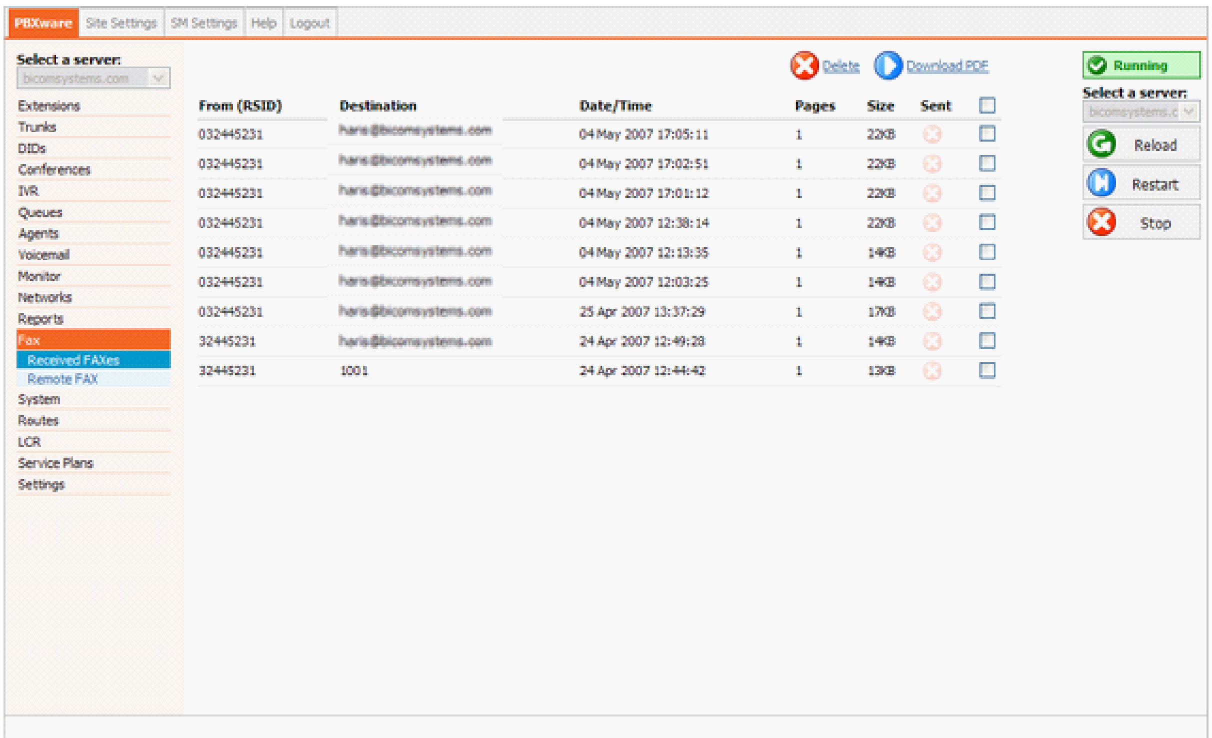
Example: AUTH LOGIN
Field Type: Display

Part XV

15 Fax

Fax window displays all faxes received by the PBXware and the ones transferred to remote systems as well

In this chapter we will cover:
- Received FAXes 276
- Remote FAX 278



15.1 Received Faxes

This screen lists all faxes received by the PBXware with the following details

Send Remotely Delete Download PDF

From (RSID)	Destination	Date/Time	Pages	Size	Sent	
032445231	Remote FAX	07 Jun 2007 13:16:44	1	6KB		
032445231	Remote FAX	07 Jun 2007 13:14:54	1	6KB		
032445231	haris@bicomsystems.com	07 Jun 2007 13:08:39	1	6KB		
032445231	haris@bicomsystems.com	07 Jun 2007 13:06:12	1	6KB		
032445231	haris@bicomsystems.com	04 May 2007 17:05:11	1	22KB		
032445231	haris@bicomsystems.com	04 May 2007 17:02:51	1	22KB		
032445231	haris@bicomsystems.com	04 May 2007 17:01:12	1	22KB		
032445231	haris@bicomsystems.com	04 May 2007 12:38:14	1	22KB		
032445231	haris@bicomsystems.com	04 May 2007 12:13:35	1	14KB		
032445231	haris@bicomsystems.com	04 May 2007 12:03:25	1	14KB		
032445231	haris@bicomsystems.com	25 Apr 2007 13:37:29	1	17KB		
32445231	haris@bicomsystems.com	24 Apr 2007 12:49:28	1	14KB		
32445231	1001	24 Apr 2007 12:44:42	1	13KB		

From:
Extension number fax was sent from
Example: 032445231
Field Type: Display

Destination:
Email address attached fax was sent to. If Extension number is displayed here, fax is sent to email address associated with the extension
Example: email@domain.con|1001
Field Type: Display

Date/Time:
Date/Time was was received
Example: 04 May 2007 12:48:12
Field Type: Display

Pages:
Number of pages in received fax
Example: 1
Field Type: Display

Size:
Fax size in KB
Example: 14KB
Field Type: Display

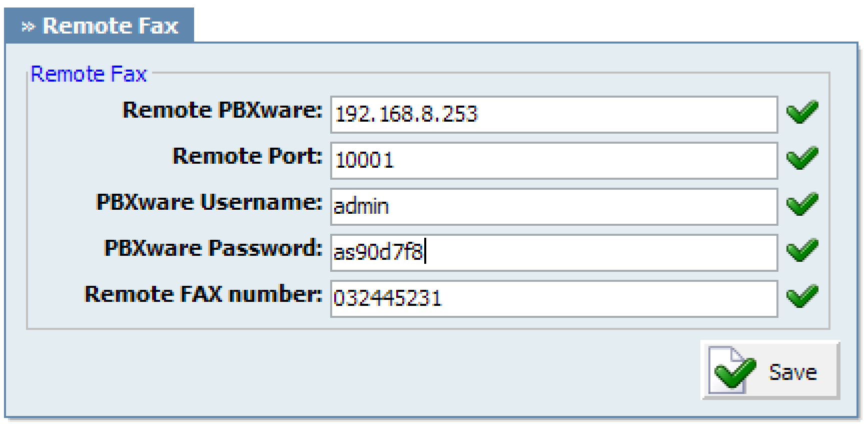
Sent:
Shows whether fax was sent remotely or not
Example: Red icon indicates local and green one indicated remote fax destination

Field Type: Display

Box used for download/delete fax actions
Example: Select this box and click 'Download' button to download selected fax
Field Type: Option button

15.2 Remote FAX

These options allow PBXware to transfer all incoming faxes to other systems. In order to do so be sure to set the following options under incoming DID ('Destination'='Fax to Email', 'Value'='remote:fax') and then provide the necessary remote system information here as follows



Remote PBXware:

IP address of remote system that is to receive fax

Example: 192.168.8.253
Field Type: [0-9]

Remote Port:

Port on remote system used for communication

Example: 10001
Field Type: [0-9]

PBXware Username:

Remote system daemon username ('Settings: Servers: Edit: Daemon Username')

Example: admin
Field Type: [0-9][a-z]

PBXware Password:

Remote system daemon password ('Settings: Servers: Edit: Daemon Password')

Example: pasd7f9
Field Type: [0-9][a-z]

Remote Fax Number:

Extension number that will be displayed as fax sender.

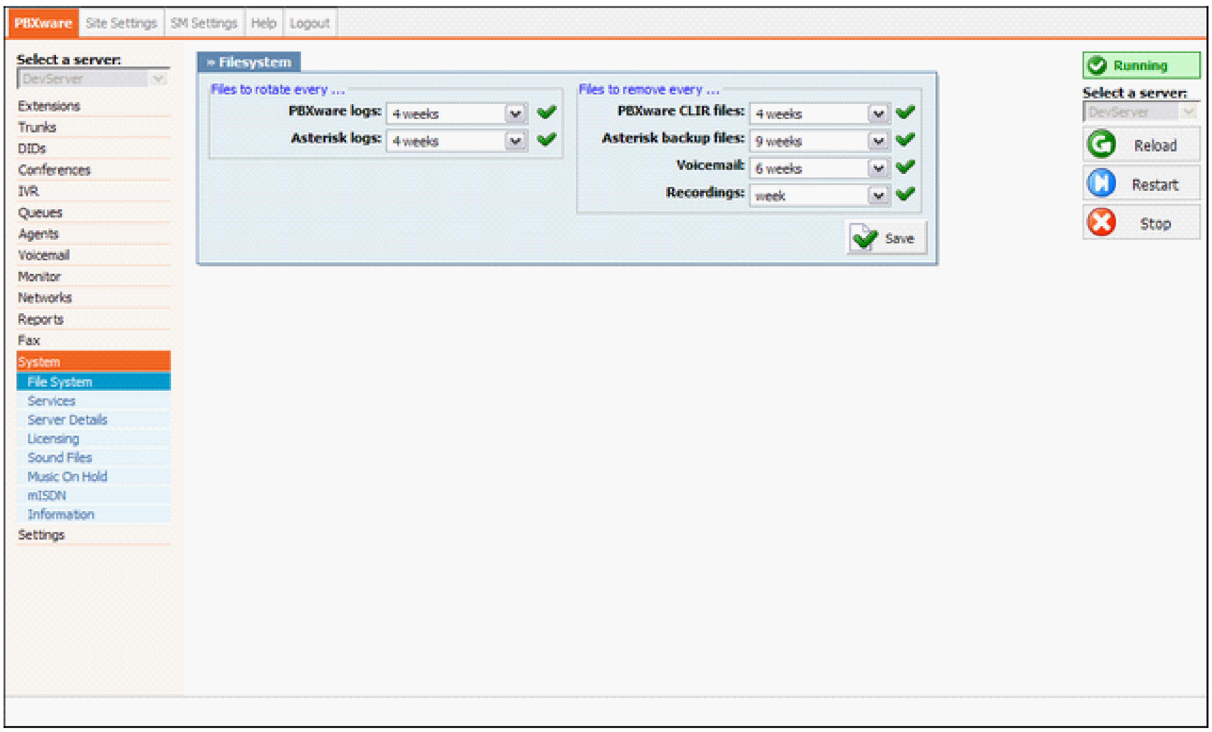
Example: 032445231
Field Type: [0-9]

Part XVI

16 System

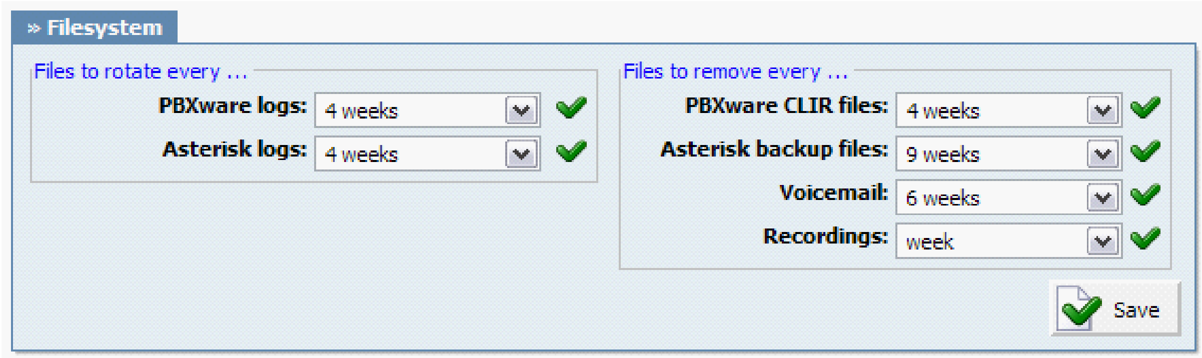
System window administers the core PBXware components such as the file system, system services, server details, licensing, sound files, MOH (Music On Hold) etc...

In this chapter we will cover:



16.1 File System

All IPBXware logs, sound recordings, CLI and CLIR files are stored on local file system. Some of these files can grow to a size which will not leave any space left on the system. This section provides management of how and when these files should be rotated or deleted in order to prevent such scenario.



Following files are rotated after selected period of time

PBXware logs:

Time PBXware logs are kept on the filesystem '/home/servers/pbxware/pw/pbxware/var/log'

Example: Default value '4 weeks'
Field Type: Select box

Asterisk logs:

Time Asterisk logs are kept on the filesystem '/home/servers/pbxare/pw/var/log/asterisk'

Example: Default value '4 weeks'
Field Type: Select box

Following files are deleted after selected period of time

PBXware CLIR files:

Time PBXware CLIR files are kept on the filesystem '/home/servers/pbxware/pw/pbxware/var/clir'

Example: Default value '4 weeks'
Field Type: Select box

Asterisk Backup files:

Time Asterisk backup files are kept on the filesystem '/home/servers/pbxware/pw/etc/asterisk/backup'

Example: Default value '9 weeks'
Field Type: Select box

Voicemail:

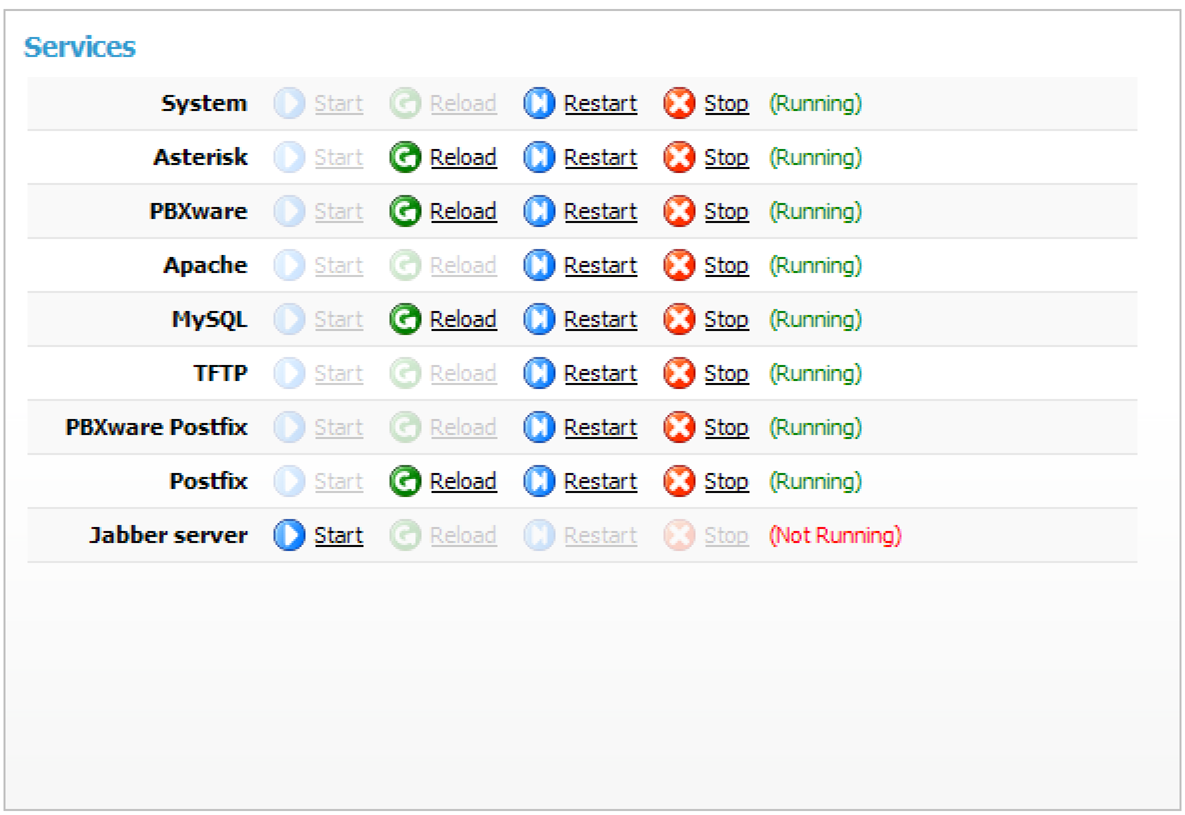
Time Voicemail files are kept on the filesystem '/home/servers/pbxware/pw/var/spool/asterisk/voicemail/default/$VOICE_MAILBOX'
Example: Default value '6 weeks'
Field Type: Select box

Recordings:

Time Recordings are kept on the filesystem '/home/servers/pbxware/pw/var/spool/ast/monitor'
Example: Default value 'week'
Field Type: Select box

16.2 Services

This window controls the basic actions (start, stop, restart, reload) of PBXware services



System:

Stop/Start/Restart the system

Example: Clicking on the 'Restart' button would reboot the system
Field Type: Command button

Asterisk:

Stop/Start/Restart/Reload the Asterisk (The core PBXware runs on)

Example: Clicking on the 'Restart' button would restart the Asterisk
Field Type: Command button

PBXware:

Stop/Start/Restart/Reload the PBXware

Example: Clicking on the 'Restart' button would restart the PBXware
Field Type: Command button

Apache:

Stop/Start/Restart/ the Apache web server

Example: Clicking on the 'Restart' button would restart the Apache web server. In case that you cannot reach the PBXware login screen this is the service you need to start/restart
Field Type: Command button

MySQL:

Stop/Start/Restart/Reload the MySQL database

Example: Clicking on the 'Restart' button would restart the MySQL database server
Field Type: Command button

TFTP:

Stop/Start/Restart/ the TFTP server

Example: TFTP is used for storing and serving the UAD/Phone auto-configuration files
Field Type: Command button

PBXware Postfix:

Stop/Start/Restart/Reload the PBXware Postfix server

Example: Clicking on the 'Restart' button would restart the PBXware Postfix server
Field Type: Command button

Postfix:

Stop/Start/Restart/Reload the Interface Postfix server

Example: Clicking on the 'Restart' button would restart the Postfix server
Field Type: Command button

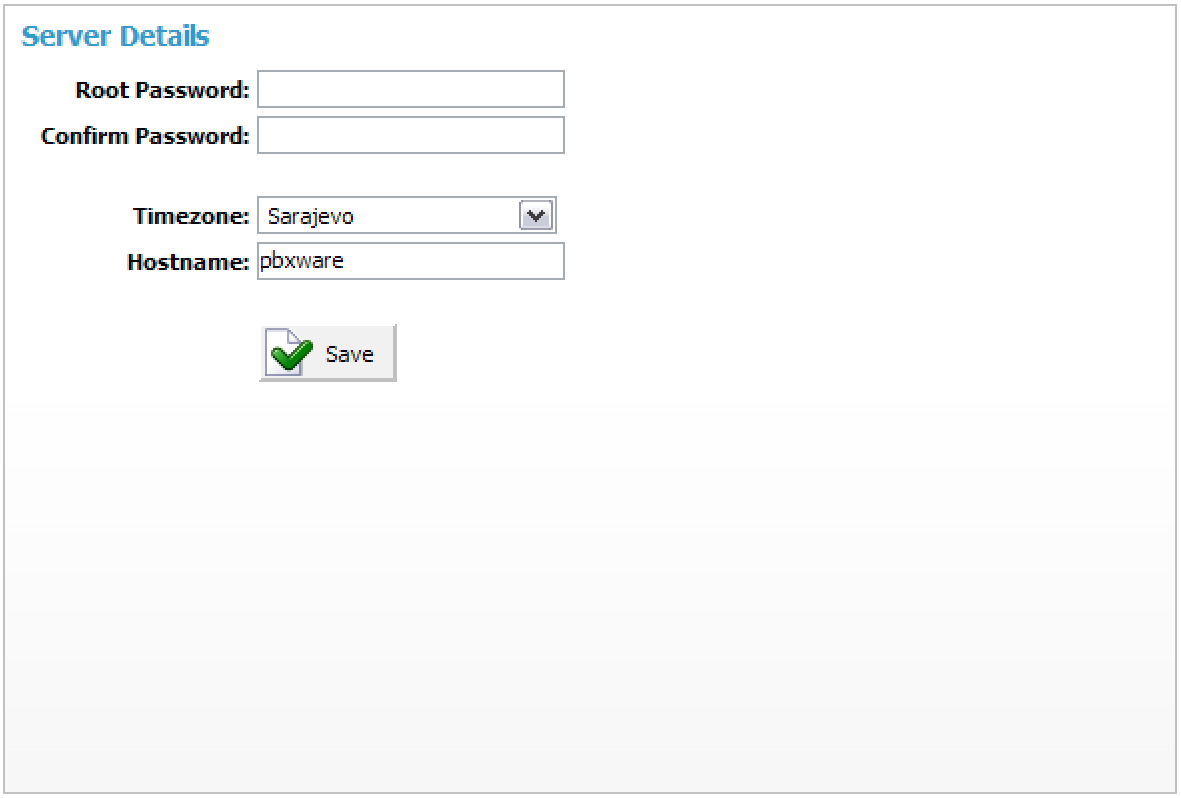
Jabber server:

Stop/Start/Restart/Reload the Jabber messaging server

Example: Clicking on the 'Restart' button would restart the Jabber server
Field Type: Command button

16.3 Server Details

This window resets the root PBXware password, timezone and hostname



Root Password:

Set the root password

Example: PBXware prompts for this password during the system/ssh login and when accessing system services throughinterface
Field Type: [a-z][0-9]

Confirm Password:

Confirm the root password

Example: Re-type the root password entered in the field above
Field Type: [a-z][0-9]

Time zone:

Time zone PBXware is located at

Example: Select the appropriate time zone, for example 'USA/East-coast
Field Type: Select box

Hostname:

The name given to machine which will identify the system on the network

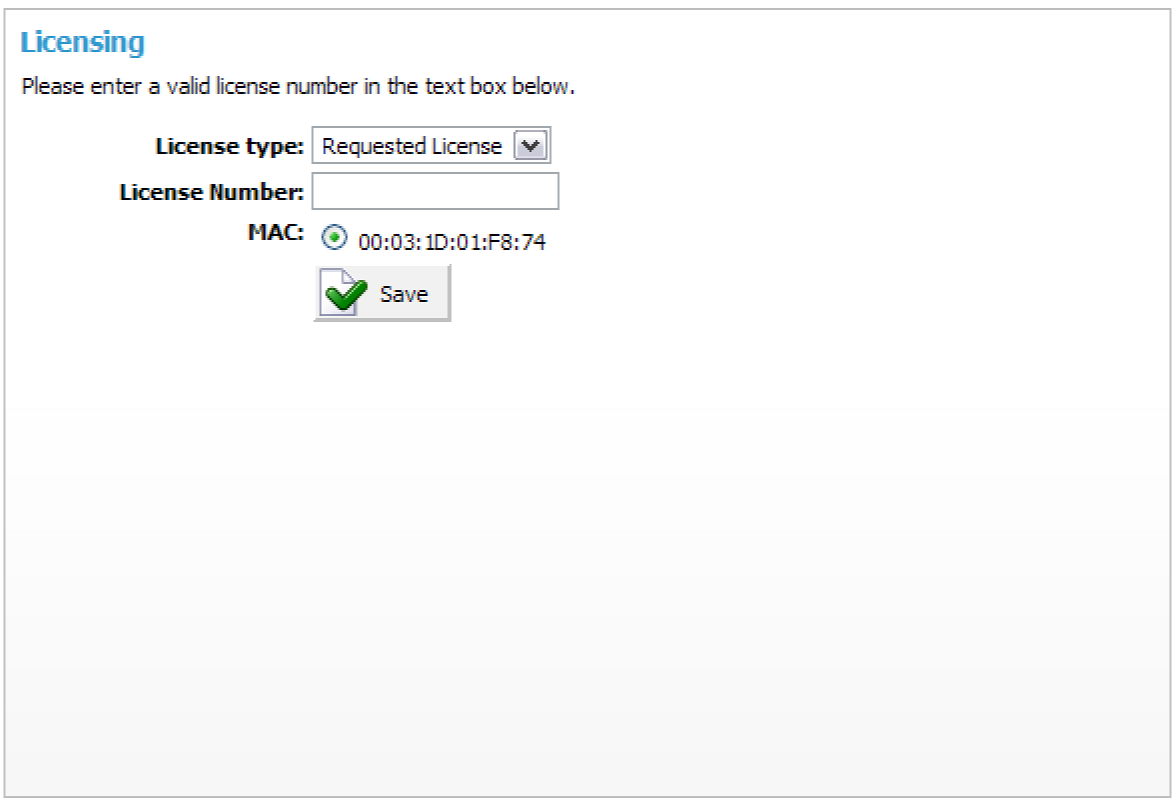
Example: hostname
Field Type: [a-z][0-9]

16.4 Licensing

This window (re)licenses the system. Free and requested licenses are available (both bond to a system MAC address).

If you are upgrading your license just paste the license key in 'License Number' field, select the MAC address assigned to a license key and click on 'Save' button

NOTE: System must have access to fully operational Internet connection in order to license the system.



Licence Type:

Select a system license type

Example: The system comes with two license types. FREE license and 'Requested License'
Field Type: Select box

License Number:

Provide system licence number as received in email.
NOTE: This field will not be active if FREE license is requested

Example: ABCDE123
Field Type: [a-z][0-9]

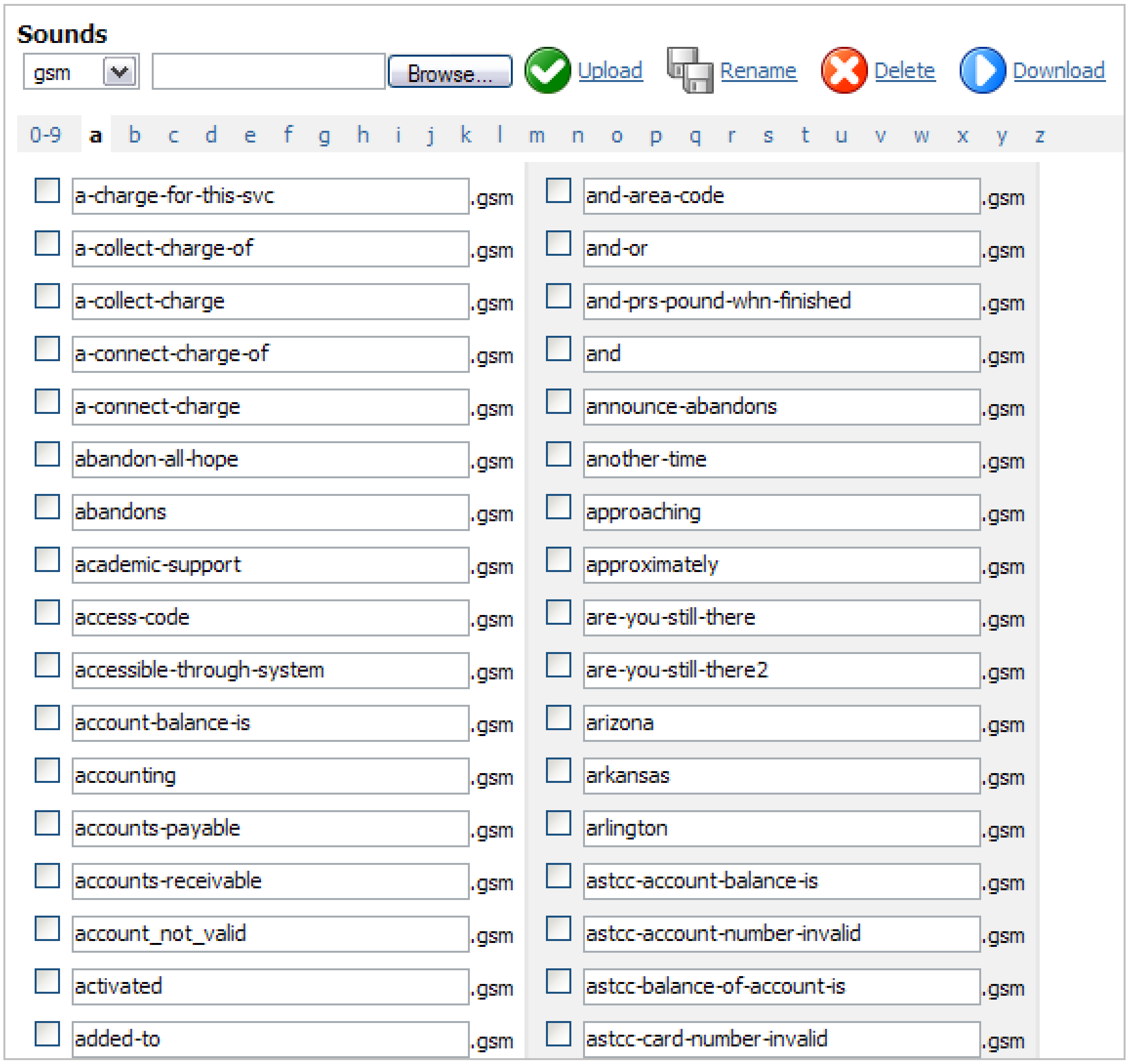
MAC:

Select a MAC address to which license will be applied

Example: If your system has multiple MAC addresses select the one you wish to assign the license to. In case this MAC address changes in the future, you will have to re-license the system
Field Type: Select options

16.5 Sound Files

This section administers all sound files used by the PBXware



Sounds:

Select among available options (gsm, ulaw, alaw, g729, ilbc, sln) to display system sound files of that type

Example: Selecting 'gsm' will display all system sounds with the .gsm file type
Field Type: Select box

Upload:

Uploads selected file from local computer to PBXware

Example: Click 'Browse' button and select a file from your desktop. Then click this button to upload selected file to PBXware
Field Type: Command button

Rename:

Renames selected sound file

Example: Select a box next to a sound file (e.g. arizona). Change 'arizona' into 'Arizona101' and click this button to rename selected sound file
Field Type: Command button

Delete:

Deletes selected sound file

Example: Select a box next to a sound file (e.g. arizona) and click this button to delete selected sound file
Field Type: Command button

Download:

Downloads selected sound file to user's desktop

Example: Select a box next to a sound file (e.g. arizona) and click this button to download selected file to your desktop
Field Type: Command button

NOTE:

PBXware will play only sound file types equal to enabled codecs on dialing extension.

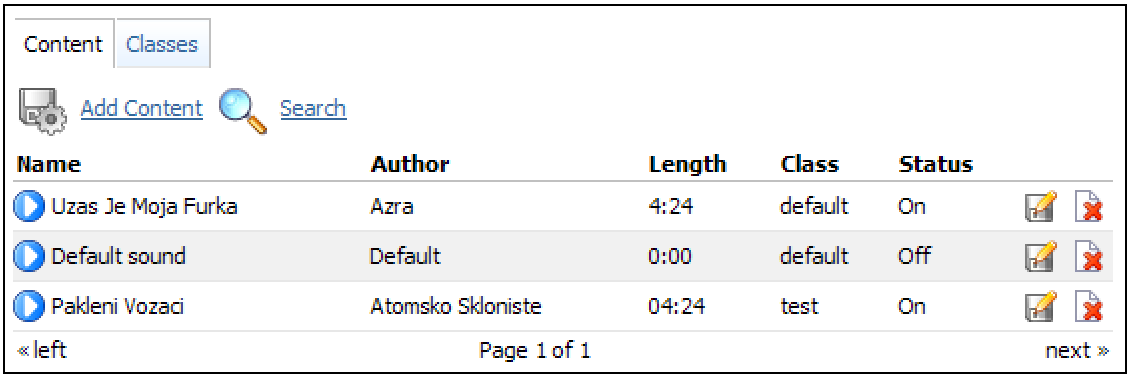
For example, Extension 1000 has only gsm codec enabled. When same Extension logs in as a Queue Agent by dialing '*202 + $AGENT_NUMBER', all sounds played by PBXware (asking for password etc...) will be in '.gsm' format.

If multiple codecs are enabled for Extension 1000(ulaw,alaw,gsm), PBXware will play sound files with better sound quality (ulaw/alaw).

16.6 Music on Hold

Music on Hold is a music or advertisements played to callers while they are waiting for agent or when put on hold for example

In this chapter we will cover:
- Content 291
- Classes 294



Content | Classes

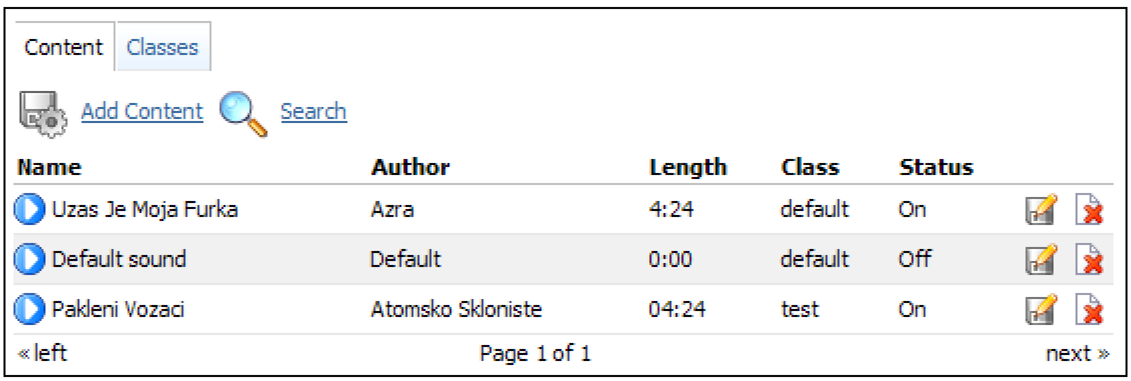
Add Content Search

Name	Author	Length	Class	Status
Uzas Je Moja Furka	Azra	4:24	default	On
Default sound	Default	0:00	default	Off
Pakleni Vozaci	Atomsko Skloniste	04:24	test	On

« left Page 1 of 1 next »

16.6.1 Content

MOH content window administers all PBXware MOH sound files. They are listed here with the following details



Content | Classes

Add Content Search

Name	Author	Length	Class	Status
Uzas Je Moja Furka	Azra	4:24	default	On
Default sound	Default	0:00	default	Off
Pakleni Vozaci	Atomsko Skloniste	04:24	test	On

« left Page 1 of 1 next »

Name:
MOH sound file name
Example: In The Club
Field Type: Display

Author:
MOH sound file author
Example: 50 Cent
Field Type: Display

Length:
MOH sound file length in min:sec
Example: 3:40
Field Type: Display

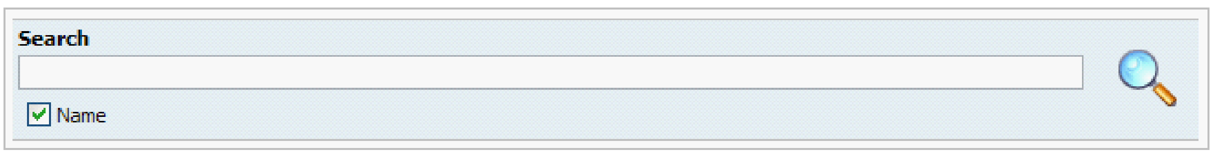
Class:
MOH class sound file belongs to
Example: Default
Field Type: Display

Status:
MOH sound file status
Example: On/Off
Field Type: Display


Edits the MOH file configuration
Example: Click to edit MOH file configuration
Field Type: Button


Deletes a MOH file from the system
Example: Click to delete a MOH file from the system
Field Type: Button

16.6.1.1 Search



Search:

Search phrase

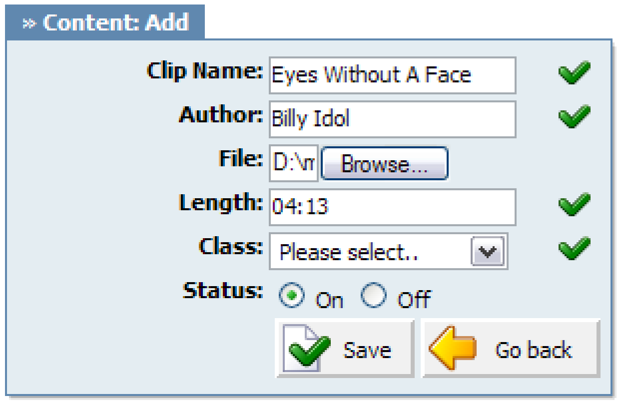
Example: Provide a search phrase here and hit enter to filter the records
Field Type: [a-z][0-9]

Name:

Should search filter be applied to track names

Example: Check the box to search track names
Field Type: Check box

16.6.1.2 Add/Edit Content



Clip name:

Audio file name

Example: Eyes Without A Face
Field Type: [a-z][0-9]

Author:

Author name

Example: Billy Idol
Field Type: [a-z][0-9]

File:

Displays full path to music file on local computer. Click 'Browse' button to select a file

Example: D:\Billy Idol\Billy Idol - Eyes Without A Face.mp3
Field Type: [a-z][0-9]

Length:

Clip length

Example: 4:13
Field Type: [0-9][:]

Class:

Select a MOH class sound file belongs to

Example: default
Field Type: Select box

Status:

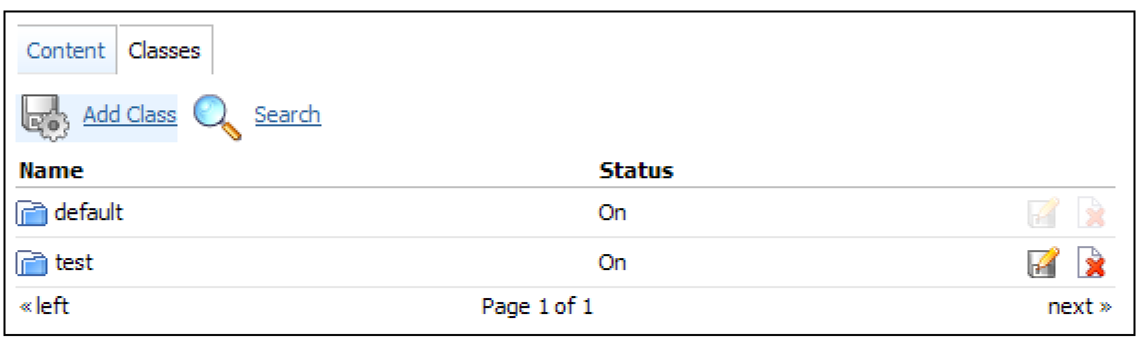
Set the status of uploaded file(active/inactive)

Example: On, Off
Field Type: Option buttons

NOTE: 'wav', 'mp3'(44100Mhz, 128bit rate), 'ogg' and 'gsm' files are allowed for upload. File size is limited to 8Mb

16.6.2 Classes

MOH classes are something like folders on a computer filled with music files. Once a MOH class is assigned to queue for example, all sound files that belong to 'default' class are played back in random order. MOH classes are listed here with the following details



Name:
MOH class name
Example: test
Field Type: Display

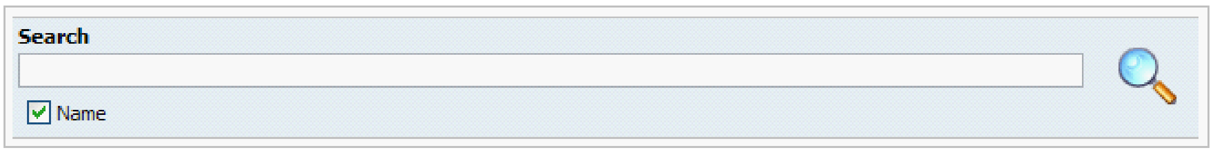
Status:
MOH class system status. When disabled, sound files that belong to the class will not be played anywhere on the system
Example: test

Field Type: Display

Edits the MOH file configuration
Example: Click to edit MOH file configuration
Field Type: Button

Deletes a MOH file from the system
Example: Click to delete a MOH file from the system
Field Type: Button

16.6.2.1 Search



Search:

Search phrase

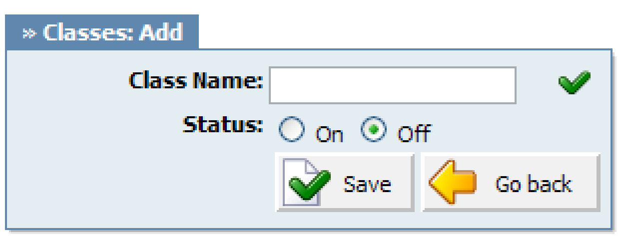
Example: Provide a search phrase here and hit enter to filter the records
Field Type: [a-z][0-9]

Name:

Should search filter be applied to class names

Example: Check the box to search class names
Field Type: Check box

16.6.2.2 Add/Edit Classes



Class Name:

MOH class name

Example: IVR greetings
Field Type: [a-z][0-9]

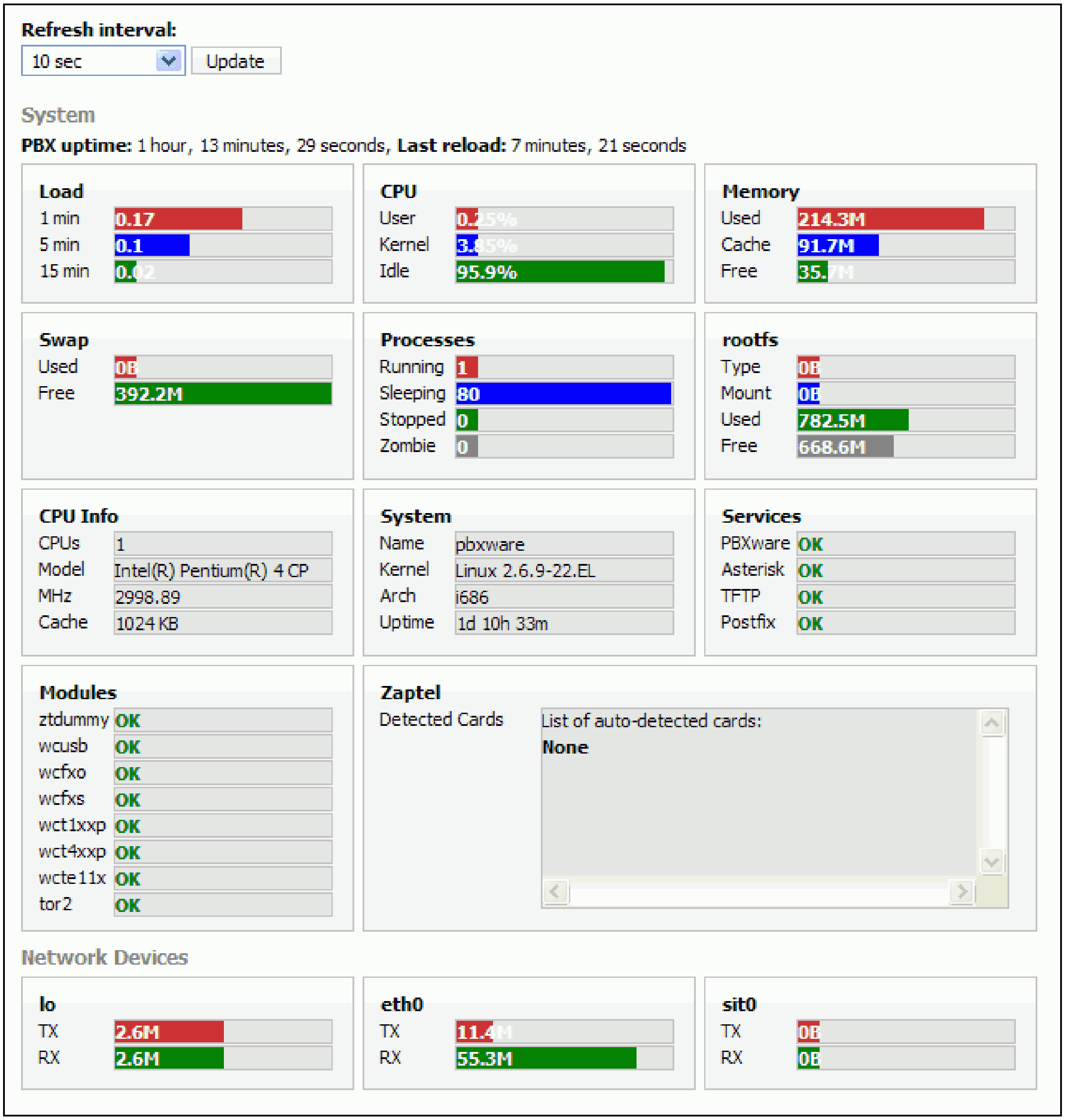
Status:

Set the class status

Example: On, Off
Field Type: Option buttons

16.7 Information

PBXware continuously monitors system services, load, ZAPTEL modules etc... in order to achieve and maintain high quality operations across the system and network. The result of these monitoring operations are also accessible to an authorized user.



Refresh Interval:
Time interval in seconds at which data is to be refreshed
Example: 10 sec
Field Type: Select box

PBXware Uptime:
Total time PBXware is available for service
Example: 1 hour, 16 minutes, 42 seconds
Field Type: Display

Last Reload:
Total time since PBXware was last reloaded
Example: 10 minutes, 34 seconds
Field Type: Display

Load:
Load shown for past 1, 5 and 15 minutes
Example: 0.16
Field Type: Display

CPU:
CPU usage by: user, kernel and idle
Example: 2.3%
Field Type: Display

Memory:
Memory usage by: Used, Cache and Free
Example: 299.1M
Field Type: Display

Swap:
Swap space usage
Example: 5.2M
Field Type: Display

Processes:
Processes by Running, Sleeping, Stopped and Zombie
Example: 1, 96, 0, 0
Field Type: Display

rootfs:
File systems present shows by type, mount, usage and free status
Example: 0B, 0B, 3.3G, 1.5G
Field Type: Display

CPU Info:
Number of CPU's, Model, Speed and Cache size
Example: Pentiun II (Deschutes), 398.28, 512KB
Field Type: Display

System:
General system details like Name, Kernel, Architecture and Uptime
Example: zenica.bicomsystems.com, Linux 2.4.21-27.0.1.EL, i686, 18d 1h 35m
Field Type: Display

Services:
Default system services running on the system
Example: PBXware, Asterisk, TFTP, Postfix
Field Type: Display

Modules:
Currently loaded ZAPTEL modules
Example: wsusb
Field Type: Display

Zaptel:
A list of all cards detected by the system is displayed here.

Channel Map displays used slots on TDM card. In this case first slot is filled with FXO module (displayed in black) while other 'Empty' slots are displayed in gray color.
Example: Channel Map: 1: FXO, 2:FXS, 3:Empty, 4:FXO
Field Type: Display

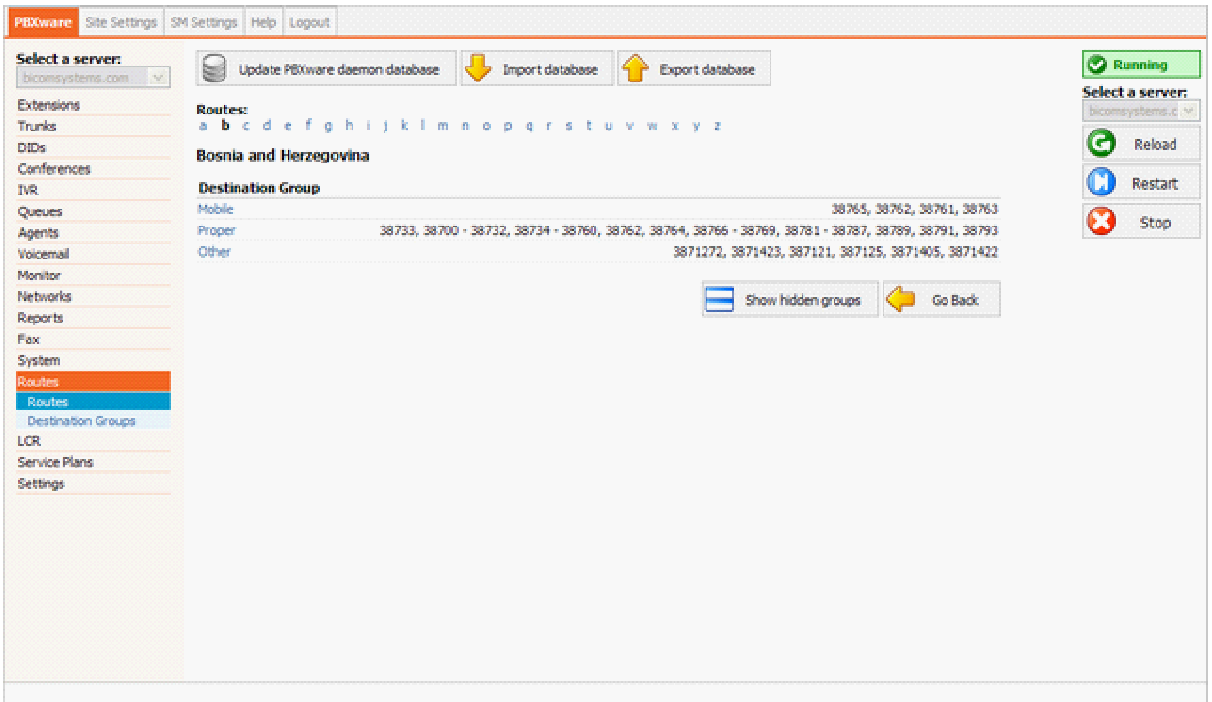
NOTE: With TDM cards, please make sure your power cable is connected. Message 'PLEASE CHECK TDM POWER CABLE' will be displayed if this happens.

Part XVI

17 Routes

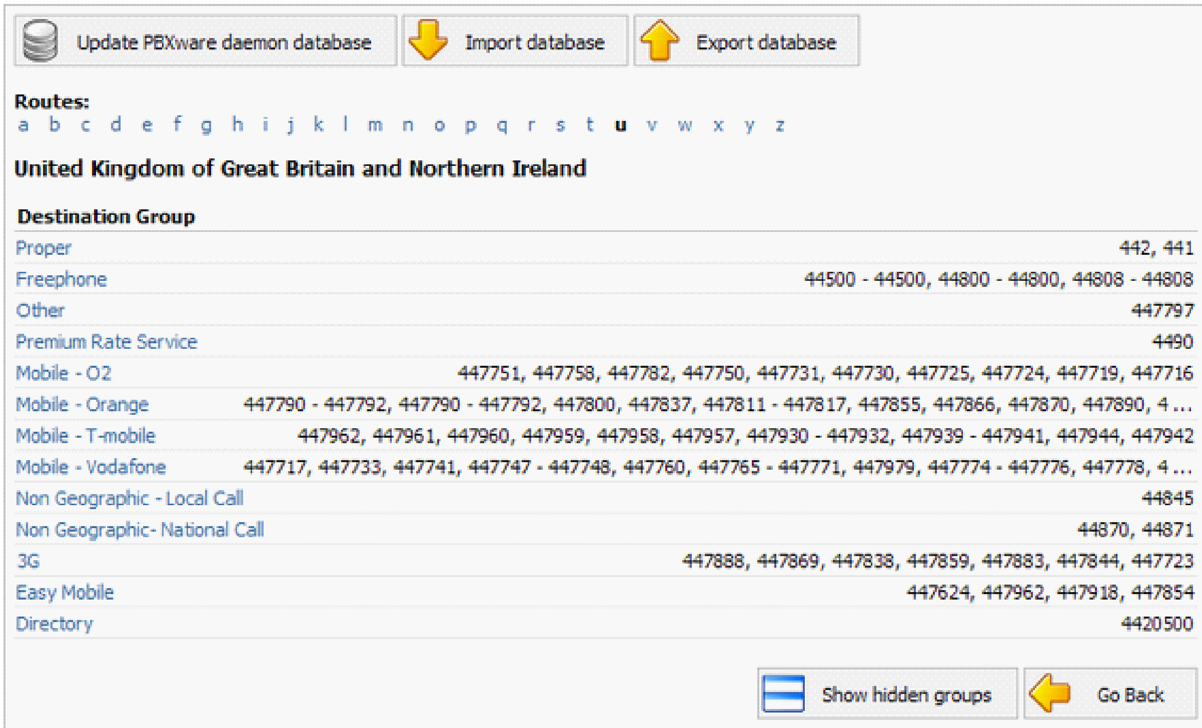
Routes identify each number dialled by users. Information is identified by the number dialled and destination group it belongs to.

In this chapter we will cover:



17.1 Routes

This window identifies all destinations by dialled numbers. All numbers provided here are in E164 format. Destinations that do not exist in this database cannot be dialed by the PBXware.



Update daemon database:

Updates daemon database with imported data

Example: Click to update
Field Type: Button

Import Database:

Imports current data from the central destination database

Example: Click to import
Field Type: Button

NOTE: Import and update operations need to be performed at regular maintenance times. In addition, any destination is not accessible the network administrator should be notified.

Export database:

Export daemon database with newly added details

Example: Click to export
Field Type: Button

Routes:

Displays alphabetically all available destination routes.

Example: A, B, C, D...
Field Type: Button

Destination group:

Further description of selected route. Displays destinations by proper, mobile etc number

Example:
- Special Service
- 48 States
- Toll Free
- Alaska
- Hawaii

Field Type: Display

Show Hidden Groups:

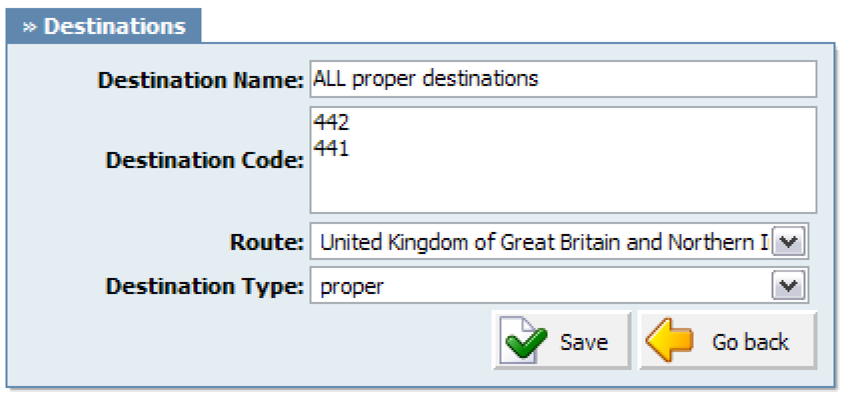
Show hidden route groups

Example: Click on this button to display all available groups
Field Type: Button

In order to view destination numbers for United Kingdom, click on 'u' letter under 'Routes' navigation. All United Kingdom destination groups will be displayed. Click on one (e.g. Proper) to view all destination numbers assign to it.

17.1.1 Add/Edit Routes

You will see these options if you have clicked on 'Add Destination' group or 'Edit' icon under routes list



Destination Name:

Unique destination name

Example: 'All proper destinations'
Field Type: [a-z][0-9]

Destination Code:

Destination code identifier

Example: Number that identifies destination (e.g. 442, where all numbers start with 442*******)
Field Type: [0-9]

Route:

Route to which destination code belongs to

Example: If route '442' belongs to United Kingdom, select 'United Kingdom' here
Field Type: Select box

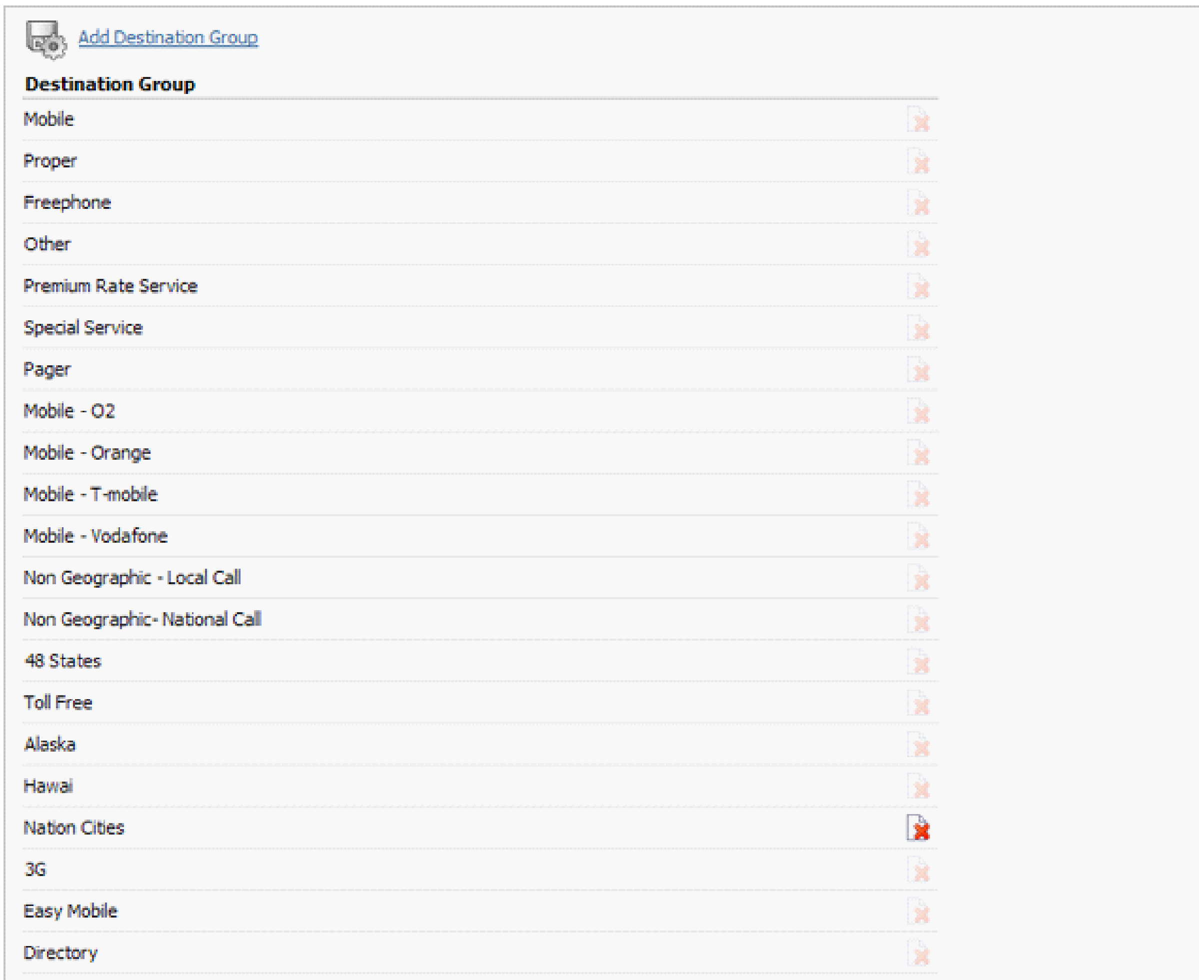
Destination Type:

Group to which destination code belongs to

Example: If route '442' belongs to proper phones, select 'Proper' here. If it belongs to mobile provider O2 for example, select 'O2' here
Field Type: Select box

17.2 Destination Groups

Each country can be assigned with many destination groups/service providers. You may add one by clicking on the 'Add Destination Group306' from this location. All groups that have no numbers assigned will not be displayed when Routes information are required



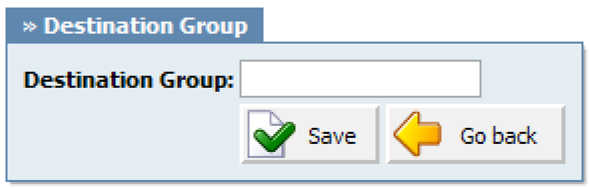
Destination Group:
Name of the destination group

Example: 48 States
Field Type: Display

Deletes a destination group from the system
Example: Click to delete a destination group from the system
Field Type: Button

17.2.1 Add/Edit Destination Group

You will see this option if you have clicked on 'Add Destination Group' option

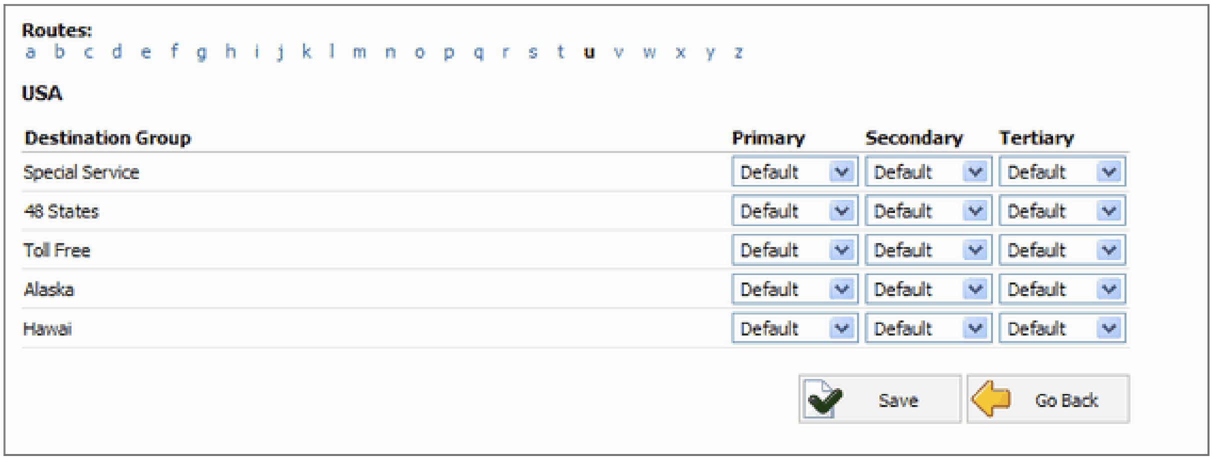


Destination Group:
Destination Group name
Example: 48 States
Field Type: [a-z][0-9]

Part XVII

18 LCR

LCR (Least Cost Routing) section allows fine tuning the systems trunks usage accordingly to the price and quality. By default, system uses default trunks for each destination in order to present this feature as simple as possible.



Routes:
Displays alphabetically all available destination routes.
Example: A, B, C, D...
Field Type: Button

Destination group:
Further description of selected route. Displays destinations by proper, mobile etc number
Example:
- Special Service
- 48 States
- Toll Free
- Alaska
- Hawaii

Field Type: Display

Primary/Secondary/Tertiary Trunk:
Select trunks to be used to desired destination
Example: Voip trunk
Field Type: Select box

Precedence

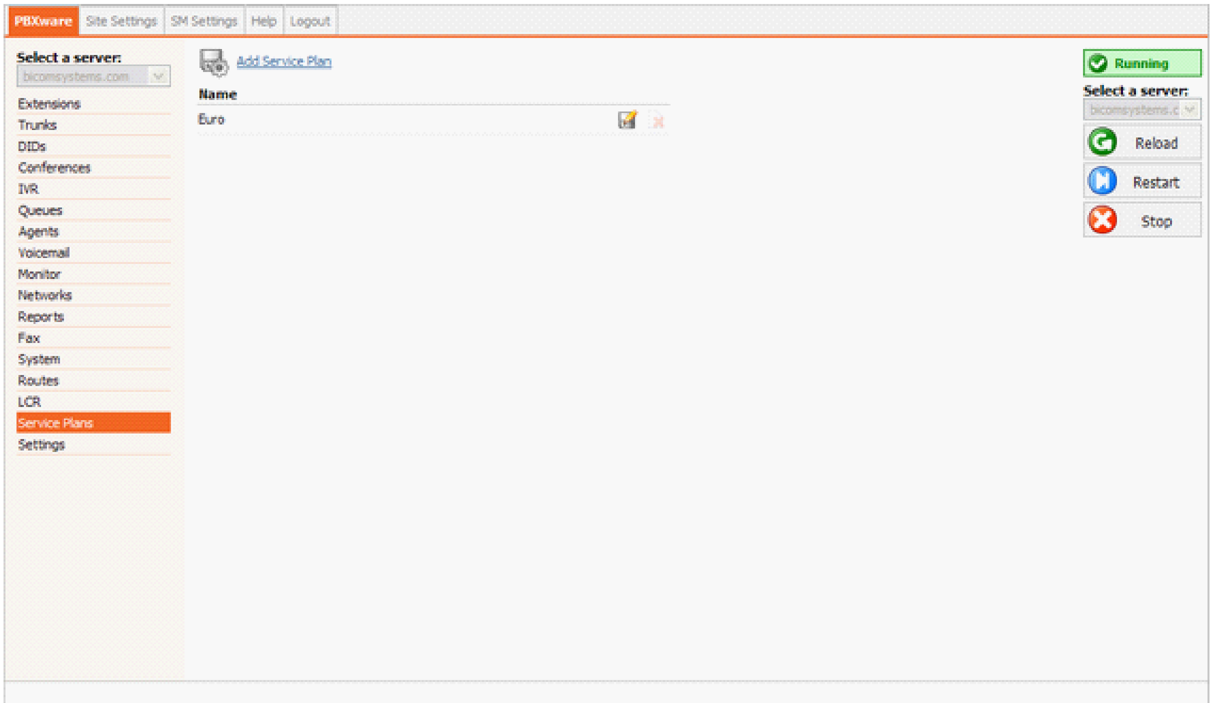
Settings:
- Default Trunks: All System calls go through trunks defined here
. MiniLCR: Overrides 'Default Trunks' and sets a specific trunk for a destination

Extensions:
- Trunks: Overrides 'Settings: Default Trunks'
- Routes: Overrides 'Settings: MiniLCR'

Part XIX

19 Service Plans

Service plan defines fare details per each destination a caller may call. This screen lists all system service plans with the following details



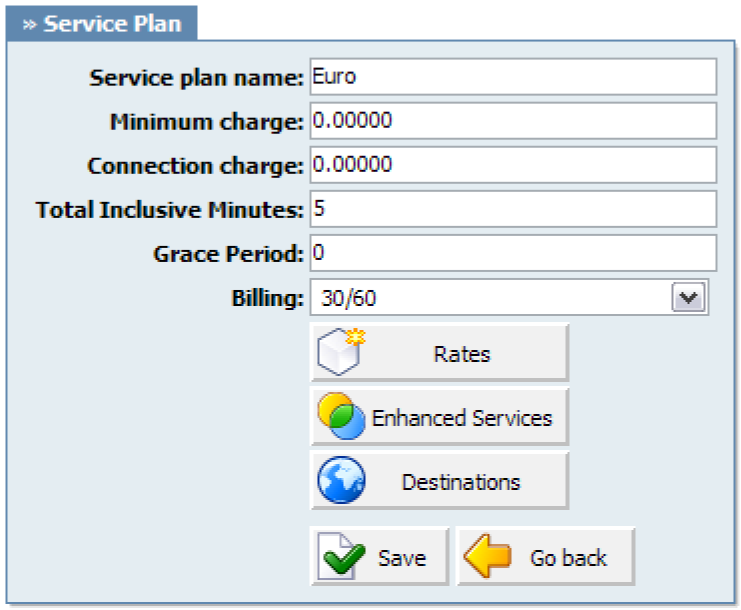
Name:
Service plan name
Example: Euro
Field Type: Display


Edits the service plan
Example: Click to edit a service plan configuration
Field Type: Button


Deletes a service plan from the system
Example: Click to delete a service plan from the system
Field Type: Button

19.1 Add/Edit Service Plan

These options fine tune the service plan with details such as minimum and connection charge, grace period and inclusive minutes



Service Plan Name:

Service plan name

Example: If service plan name is 'Euro', select this name under 'Service Plan' under extensions to apply it to that Extension
Field Type: [a-z][0-9]

Minimum charge:

Minimum charge applied to each made call regardless of the call duration

Example: If call is made, no matter how much it lasts, this 'minimum charge' will be applied
Field Type: [0-9]

Connection charge:

Charge applied to any call that leaves the system (regardless if other party answers or not)

Example: If this charge is set to 0.4, each call that leaves the system will be charged that amount regardless if other party answers the call or not
Field Type: [0-9]

Total Inclusive Minutes:

Total number of inclusive minutes assigned to a service plan

Example: If this field is set to '5', each user assigned to this service plan will have 5 free minutes of call time to any location.
NOTE: 'Minimum charge' and 'Connection charge' will be applied if set.
Field Type: [0-9]

Grace Period:

Number of seconds at the beginning of a call that are not charged

Example: If grace period is set to 10, and call lasted 15 seconds, only 5 seconds of the call will be charged
Field Type: [0-9]

Billing:

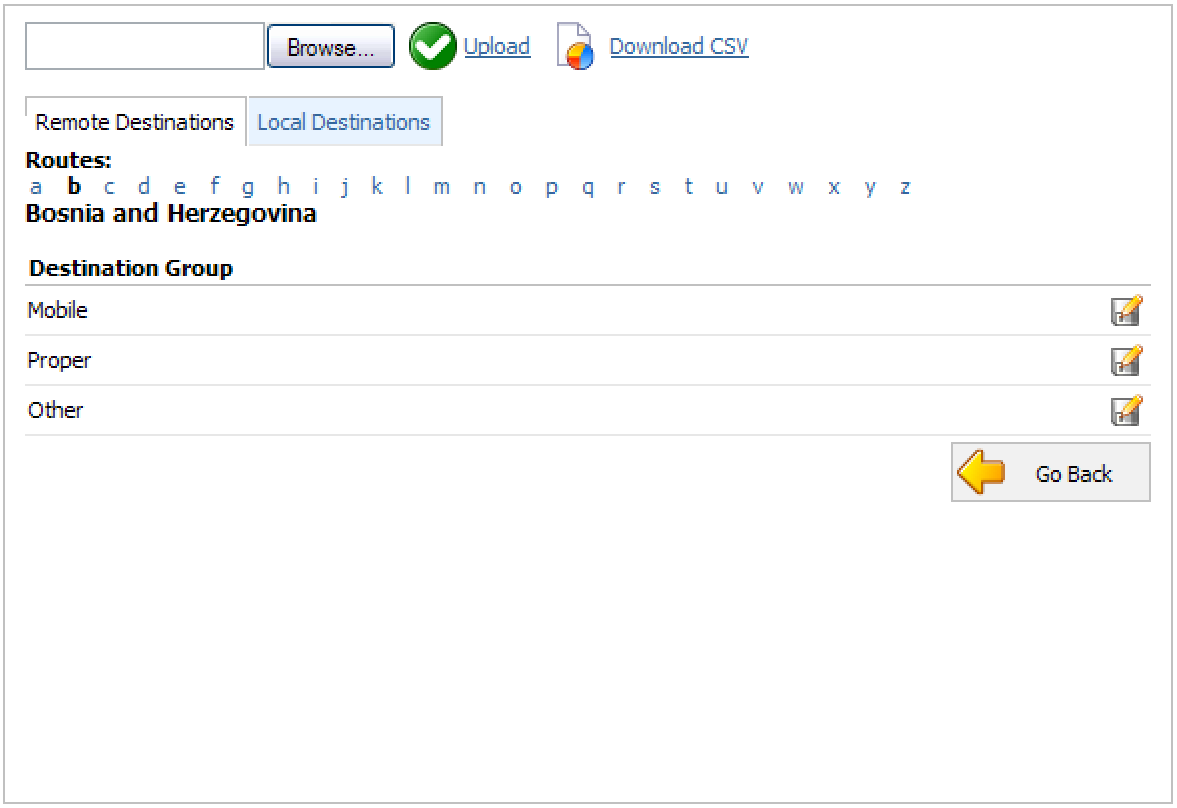
Billing type (MINIMUM_CHARGED/CHARGE_EVERY_$SECONDS)

Example: If '30/6' is selected, and you've made the call which lasted 12 seconds, it will be billed as if you've made a 30 seconds call. If the call lasted for 39 seconds - it will be billed as if the call lasted 42 seconds (30 + 6 + 6 = 42)
Field Type: Select box

19.1.1 Rates

These options set incoming/outgoing rates per each known destination

In this chapter we will cover:
- Upload/Download 314
- Routes 315



Destination Group:
Destination Group name
Example: Mobile
Field Type: Display

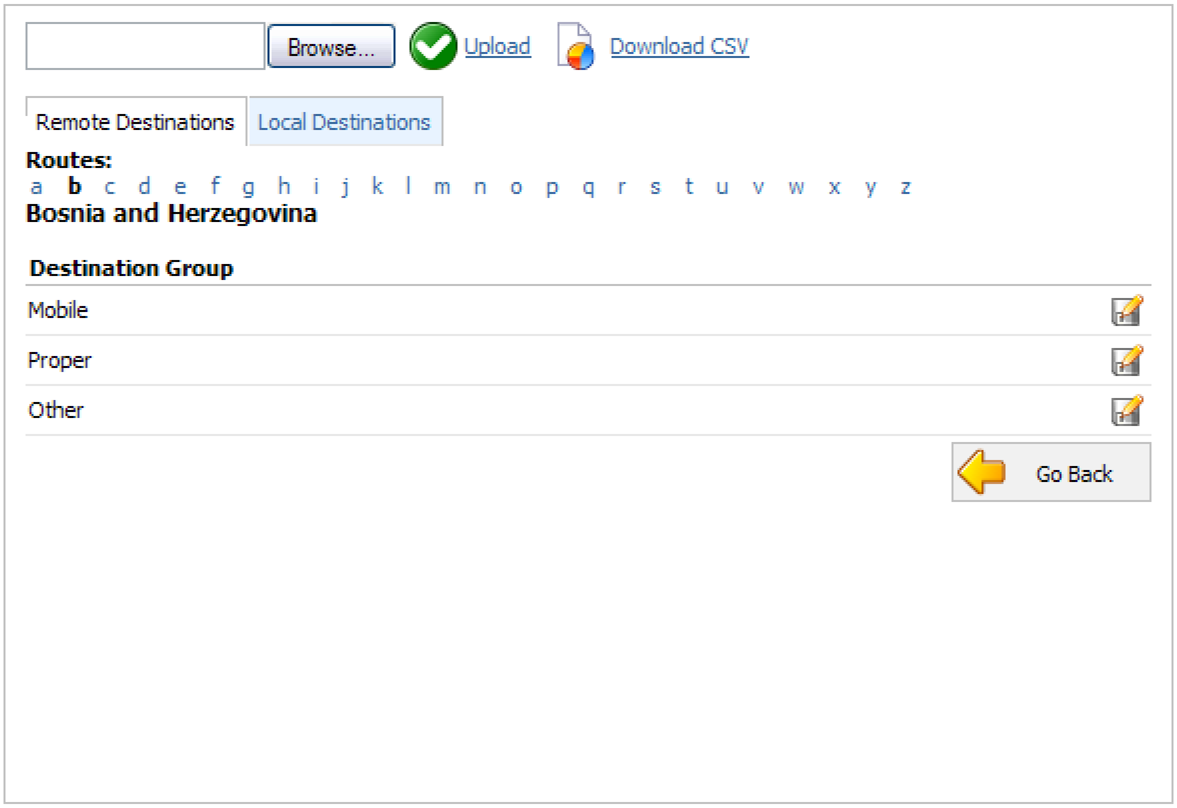
Edits the Destination Group charges
Example: Click to edit a Destination Group charges
Field Type: Button

19.1.1.1 Upload/Download

Destination Group rates can be easily uploaded and download from the server. Update the .CSV file on your desktop, click on 'Browse' button, select the file and click on 'Upload' button.

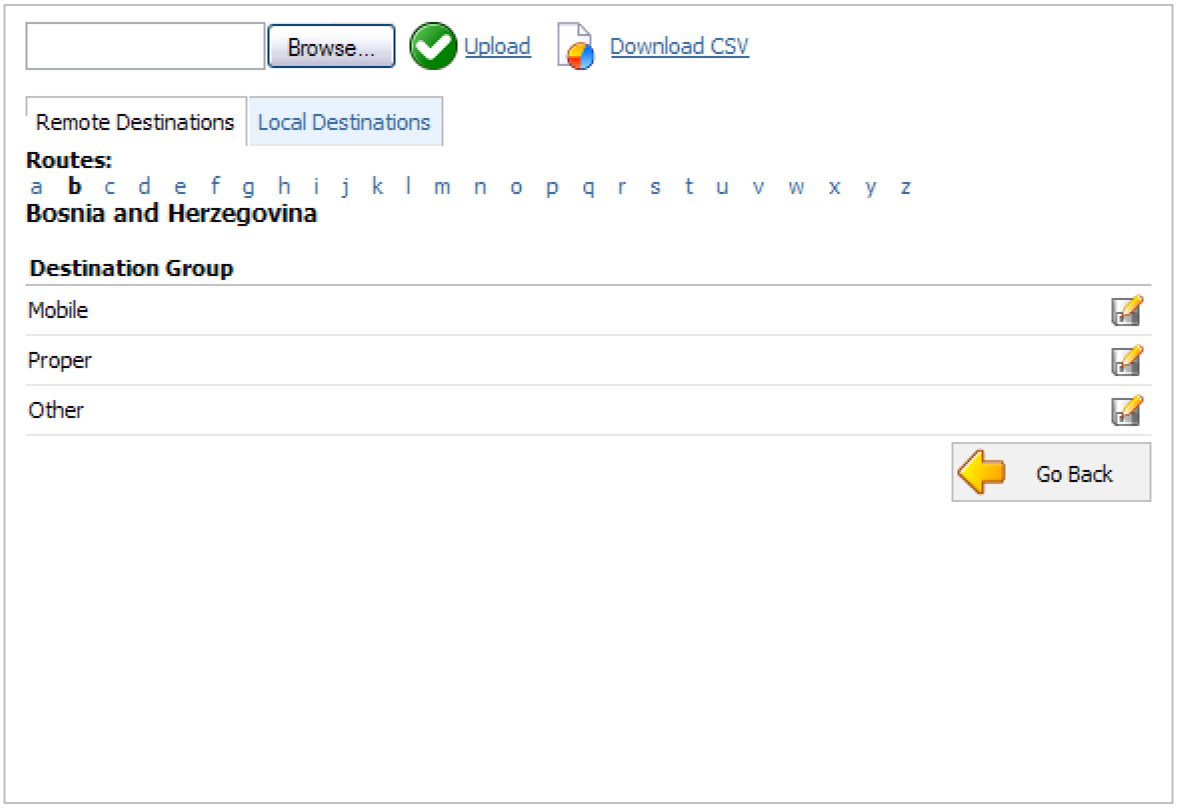
NOTE: CSV file must be in following format (Code,"Route","Destination","Outbound","Inbound"). For example 93,"Afghanistan","mobile"

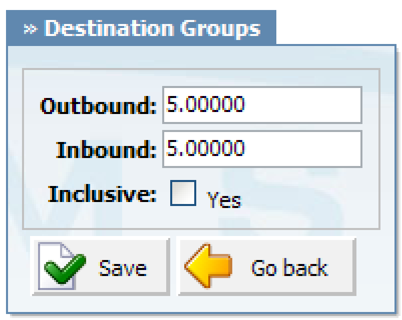
To download rates file from the server just click on 'Download CSV' button.



19.1.1.2 Routes

Click on a letter under a 'Routes' navigation 'B' for example and select 'Bosnia and Herzegovina'. A list of 'Destination Groups' will be displayed under 'Destination Group'. Click on a 'Edit' button to edit the Destination Groups charges.





Outbound:

Destination group outbound charge

Example: If you have edit the 'Mobile' destination group and set this options to 5.00000, this rate will be applied to all made calls to Mobile destination
Field Type: [0-9]

Inbound:

Destination group inbound charge

Example: If you have edit the 'Mobile' destination group and set this options to 5.00000, this rate will be applied to all received calls from this Mobile destination
Field Type: [0-9]

Inclusive:

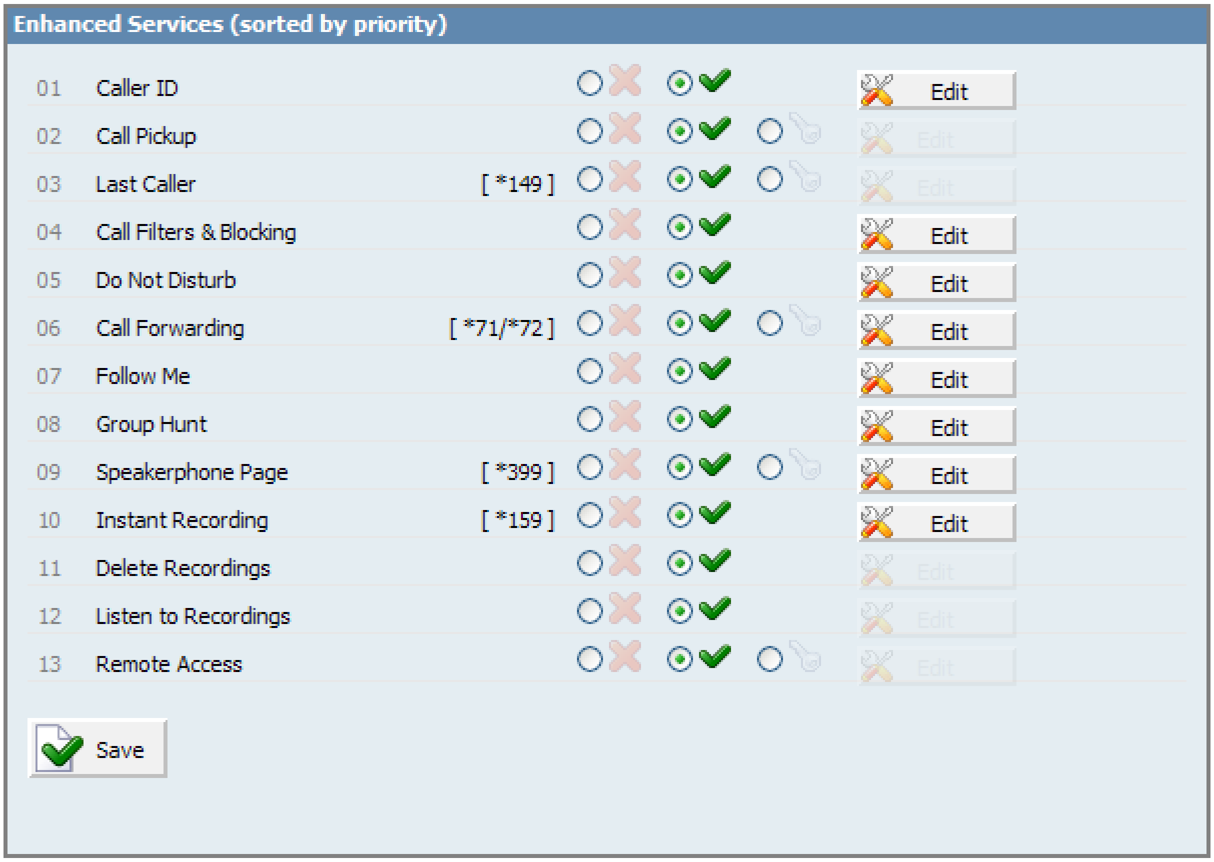
Should inclusive minutes be calculated for this destination

Example: If this option is enabled, inclusive minutes will be applied when dialling of receiving calls from this location
Field Type: Option buttons

19.1.2 Enhanced Services

Enhanced Services set here will be applied to all users assigned with this Service Plan. For example, if 'Euro' Service Plan is set to have only 'Call Forwarding' enabled, all users with 'Euro' Service Plan will have 'Call Forwarding' enabled only.

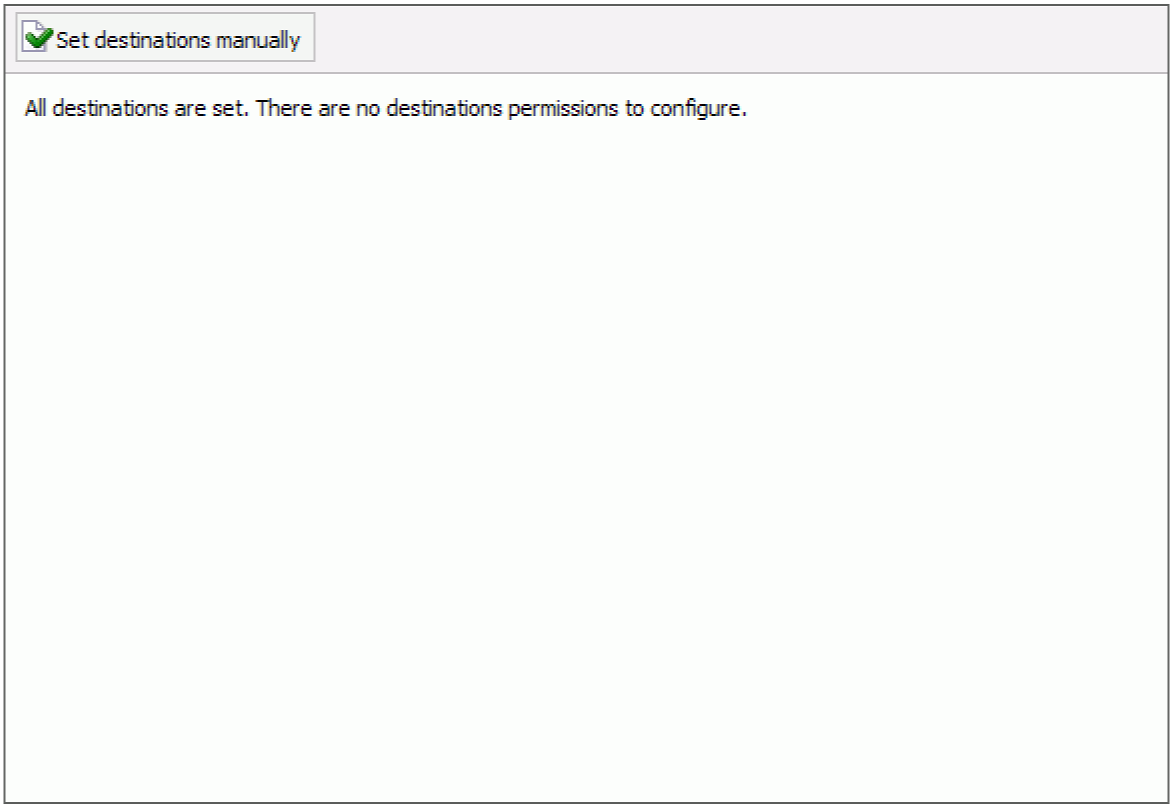
For more on Enhanced Services please click here 72

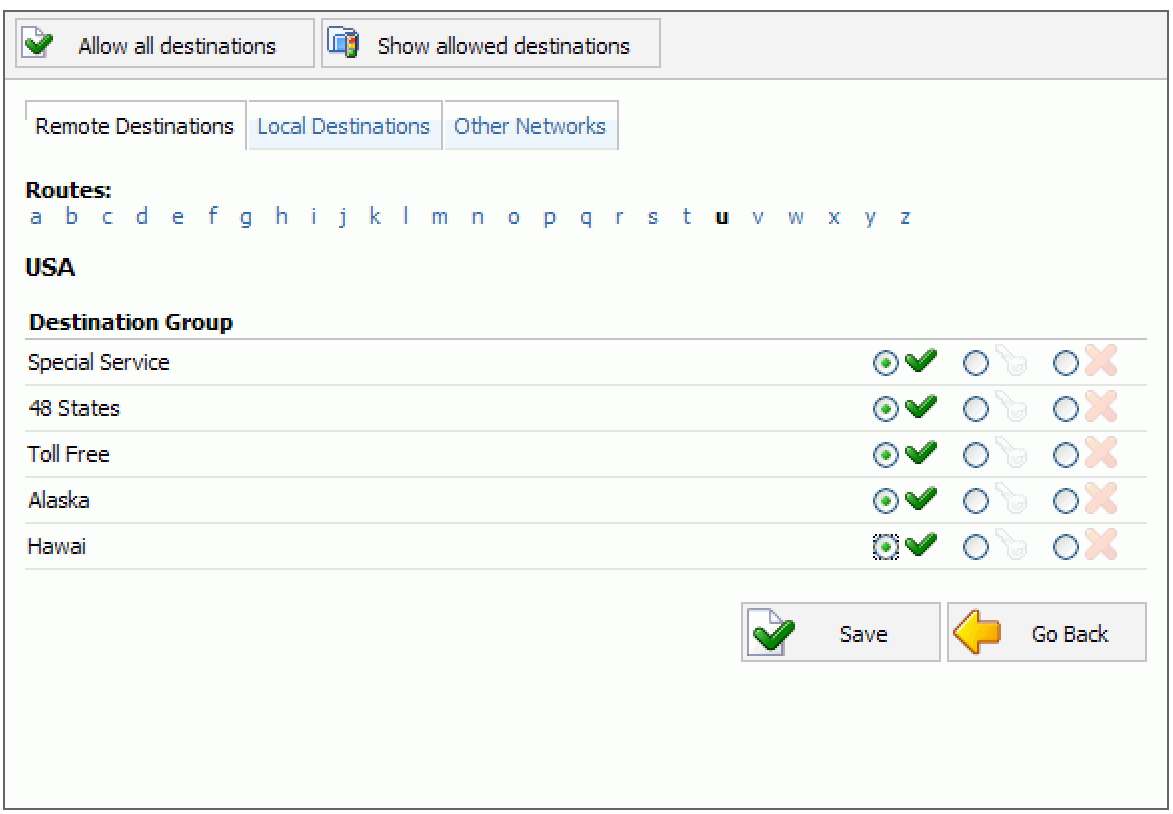


19.1.3 Destinations

Destinations set here will be applied to all users assigned with this Service Plan. For example, if 'Euro' Service Plan is set to have 'UK: Proper' destination allowed only, all users with 'Euro' Service Plan will be able to call only 'UK: Proper'

For more on Destinations please click here 68


Set destinations manually
All destinations are set. There are no destinations permissions to configure.


Allow all destinations
Show allowed destinations
Remote Destinations
Local Destinations
Other Networks
Routes:
a b c d e f g h i j k l m n o p q r s t u v w x y z
USA
Destination Group
Special Service
48 States
Toll Free
Alaska
Hawai
Save
Go Back

Part XX

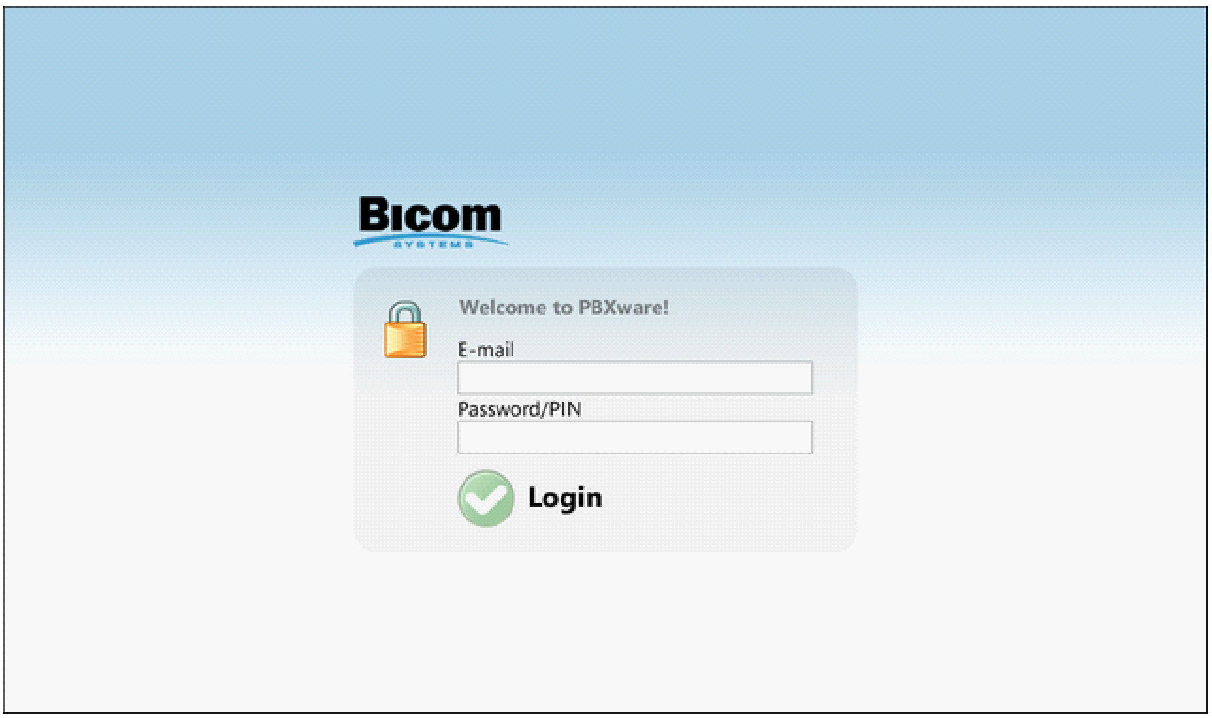
20 Self Care

Self Care is extension administration interface used by the extension owner

In this chapter we will cover:

20.1 Login

In order to login into Self Care point your browser to: http://$IPADDRESS/ (For Example: http://192.168.1.1/)



Email:

Email address assigned to extension

Example: Provided email address is used as a username for logging into the Self Care (e.g. email@example.com)

Field Type: [a-z] [0-9] [@_.-]

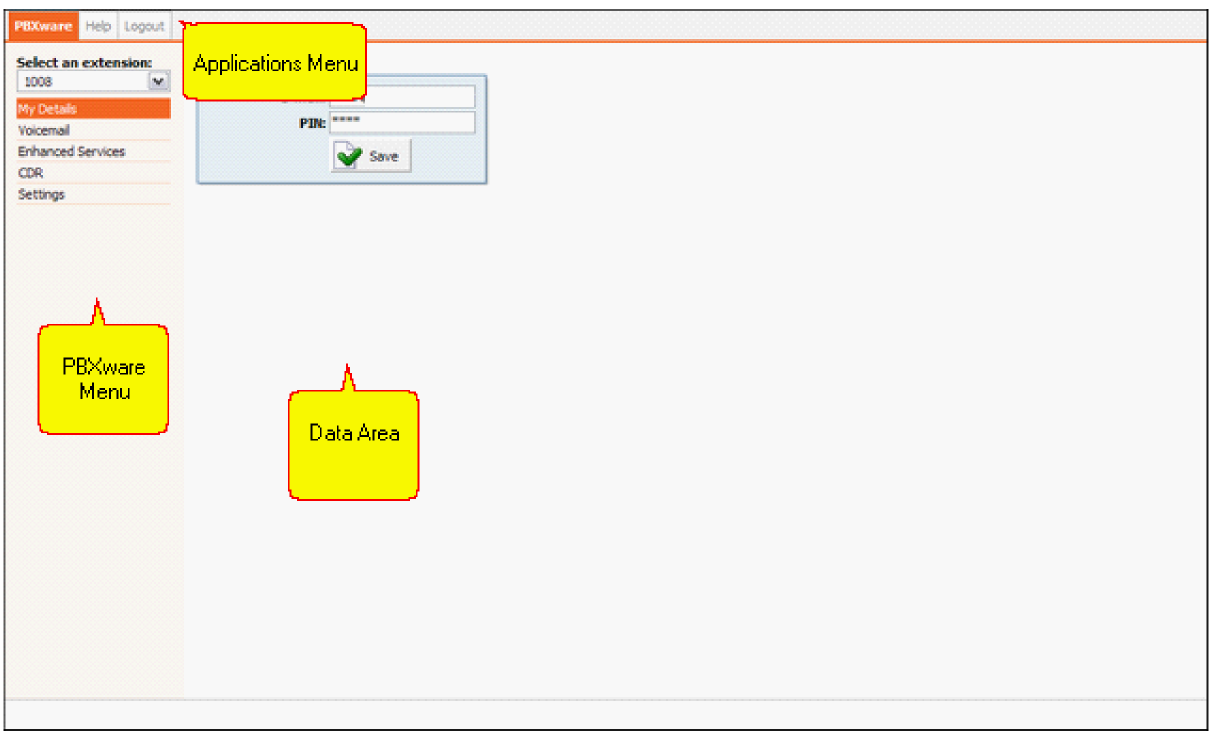
Password / PIN:

PIN assigned to extension

Example: This field accepts extension PIN (e.g. 1981)
Field Type: [0-9]

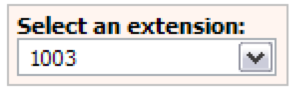
20.2 Administration Interface

In this chapter we will cover:
- Extension Control 323
- My Details 324
- Voicemail 325
- Enhanced Services 328
- CDR 342
- Settings 347

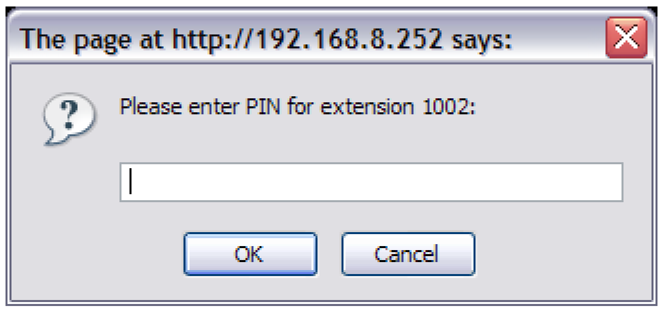


20.2.1 Extension Control

User can monitor multiple extensions through Self Care interface. To administer different extension select its network number from the 'Select an extension' select box.



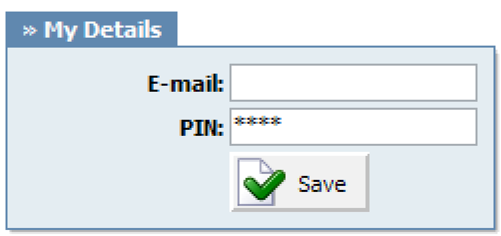
You will be asked to authenticate by providing extension PIN number. If correct extension PIN is provided, user will administer selected extension.



20.2.2 My Details

User can manage his email account and PIN associated with his extension.

NOTE: If user has voicemail account only(no system extension), this feature will be disabled.



E-mail:

E-mail address associated with the extension. This address is used for various system notifications and for user logging into Self Care

Example: To login into Self Care type this email address into 'E-mail' field
Field Type: [a-z] [0-9] [@._-]

PIN (Personal Identification Number):

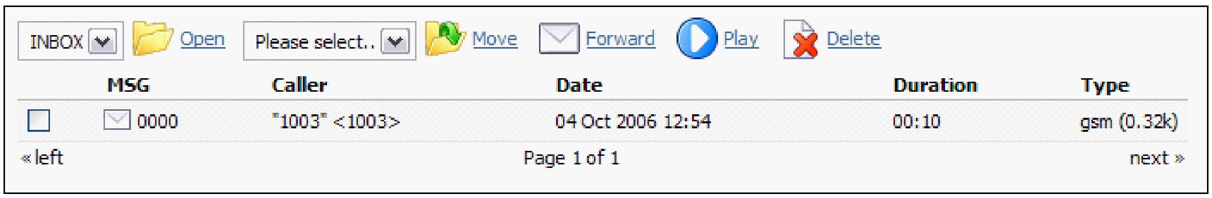
Four digit password used for accessing voicemail and other additional PBXware services as well as logging into Self Care

Example: To login into Self Care type this number into 'PIN' field
Field Type: [0-9]

20.2.3 Voicemail

User can manage voice messages left on his extension from this location.

In this chapter we will cover:
- Actions 326
- Voicemail Options 327



Msg:
Voicemail message identification number
Example: 0000
Field Type: Display

Caller:
Identifies the user who has left the message by his name and extension number
Example: "BobReilly" <5000>
Field Type: Display

Date:
Time/Date a voicemail has been received in inbox
Example: 13 Apr 2006 15:12
Field Type: Display

Duration:
Time duration of voice message
Example: 00:18
Field Type: Display

Type:
Voicemail file type and size
Example: wav49 (9.07k)
Field Type: Display

NOTE: Disk Space Used By Voicemail Recording
With continuously tone 60 seconds:
- wav49 = 91.0kb

- wav = 863.0kb
- gsm = 91.0kb

With continuously silent tone (without sound) 60 sec:
- wav49 = 0.38kb
- wav = 3.0kb
- gsm. = 0.32k b

20.2.3.1 Actions

Open:

Displays the content of a voice inbox folder

Example: Select a destination folder in a Select box and click this button to display its contents
Field Type: Command Button

Move:

Moves the content into different voice mailbox location/directory

Example: Select the box next to a voice message, set the destination folder in the Select box and click this button to move the voice message to new destination/folder
Field Type: Command Button

Forward:

Forward the voice message to other network extension inbox

Example: Select the box next to a voice message and click this button. When prompted for extension, type '1005' for example and selected voice message will be transferred to the voice inbox of the network extension 1005
Field Type: Command Button

Play:

Downloads/Plays the voice message

Example: Select the box next to a voice message and click this button to download the message on the Desktop or to play it in favorite media played (depending on the option selected in the popup window)
Field Type: Command Button

Delete:

Deletes the voce message from the inbox

Example: Select the box next to a voice message and click this button to permanently delete the voice message from the inbox
Field Type: Command Button

20.2.3.2 Voicemail Options

Once the user is transferred to party's voice box 'Please leave a detail message after the tone. If you would like to speak to the operator, press 0' message will be heard.

User has two options:
1. To leave a voice message that is ended by pressing # key or by hanging up, or
2. To reach an operator by dialing 0

If 0 is dialed 'Press 1 to accept this recording, otherwise please continue to hold' message will be heard.

User has two options:
1. Press 1 to save your message and dial the operator. 'Please hold while i try that extension' message played.
2. Continue to hold to delete your message and dial the operator. 'Message deleted, please hold while i try that extension' message played.

Standard voicemail options with all voicemail settings set to 'Yes':

1 Read voicemail messages
2 Change folders
- **0** Mailbox options
- **1** Record your unavailable message
- **2** Record your busy message
- **3** Record your name
- **4** Record your temporary message (new in Asterisk v1.2)
- **5** Change your password
- ***** Return to the main menu

3 Advanced options (with option to reply; introduced in Asterisk CVS Head April 28, 2004 with 'enhanced voicemail')
- **1** Reply
- **2** Call back(1)
- **3** Envelope
- **4** Outgoing call(1)

4 Play previous message
5 Repeat current message
6 Play next message
7 Delete current message
8 Forward message to another mailbox
9 Save message in a folder
***** Help; during msg playback: Rewind
Exit; during msg playback: Skip forward

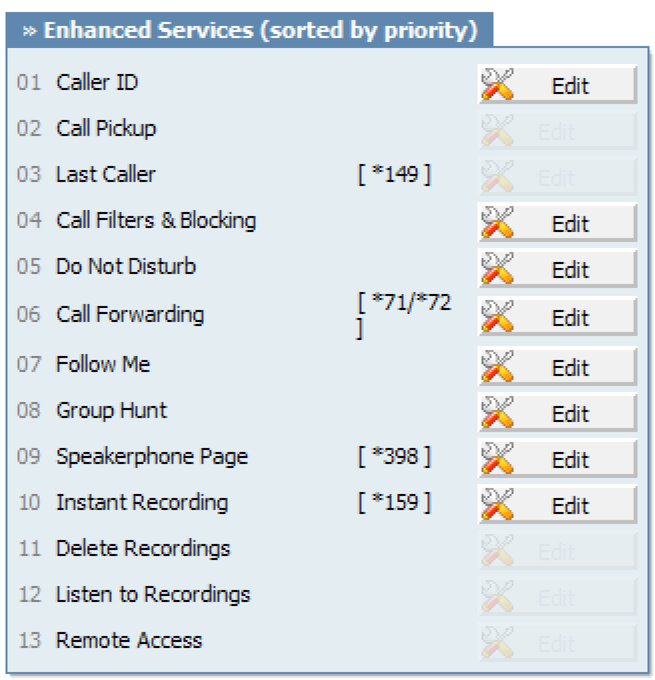
***** ***** Help
***** **#** Exit

After recording a message (incoming message, busy/unavailable greeting, or name)
1 Accept
2 Review
3 Re-record
0 Reach operator(1) (not available when recording greetings/name)

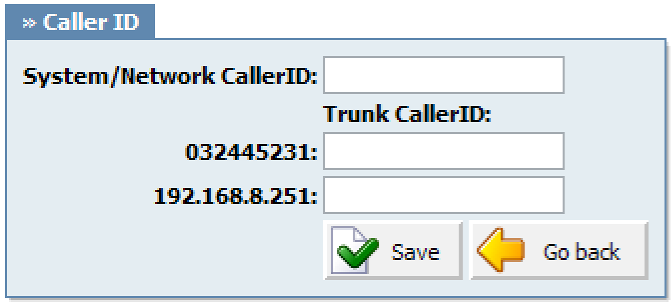
20.2.4 Enhanced Services

In this chapter we will cover:

NOTE: If user has voicemail account only, and no system extension, this feature will be disabled.



20.2.4.1 Caller ID



System/Network Caller ID:

Information provided here will replace default Caller ID information on System/Network level

Example: Set this option to 'JaKe 2007' and call extension 1005 for example. On the display of the 1005 UAD/Phone, 'JaKe 2007' will be displayed as a caller id information on system/network level
Field Type: [a-z][0-9]

Trunk CallerID:

Caller ID information set per trunk

Example: For each trunk available on the system there will be a field in which you may set custom caller id data that is used when calling over that specific trunk
Field Type: [a-z][0-9]

20.2.4.2 Call Pickup

Call Pickup:

This service enables user to pickup ringing calls of the same call group.

Example: Dial '*8' to pickup a call from the same call group, or '*88 + $EXTENSION' to pickup calls from different call groups
Field Type: Option buttons

Example:

Extension A:

Call Group = 1
Pickup Group = 3,4

Extension B:
Call Group = 2
Pickup Group = 1

- If A is ringing, B can pickup the ringing call by dialing '*8'.
- If B is ringing, A cannot pickup the ringing call because B's call group = 2, and A can pickup only call groups 3,4

NOTE: Grouping works only within a technology(SIP to SIP or IAX to IAX).

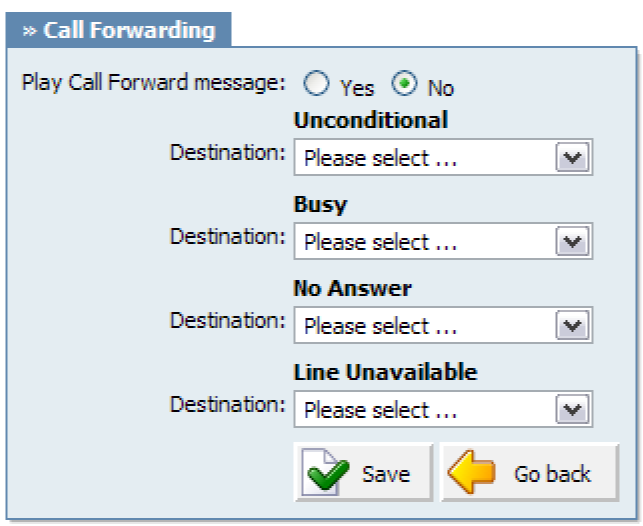
20.2.4.3 Last Caller

Last Caller:

This service will dial the last extension that was calling you. For example, dial '*149' to hear the extension number and then press '0' to call that number

20.2.4.4 Call Filters & Blocking

This service forwards calls to other extensions depending on extension response/status



Play Call Forwarding Message:

Notify users of a transfer being made.

Example: If this option is set to 'Yes', the caller would hear a 'Please wait, your call is being forwarded. You're not being charged for the forwarding part of the call' message, notifying him that a transfer is being made.
Field Type: Option buttons

Unconditional:

Forward all incoming calls.

Example: Calls can be forwarded to other extension numbers(Local or Remote) and to local Voicemail Boxes.
Field Type: Option buttons

Destination:
Field Type: [0-9]

Busy:

Forward all incoming calls if the extension is busy.

Example: Calls can be forwarded to other extension numbers(Local or Remote) and to local Voicemail Boxes if the line has reached the maximum incoming calls limit.
Field Type: Option buttons

Destination:
Field Type: [0-9]

No Answer:

Forward all incoming calls if the extension doesn't answer the incoming call.

Example: Calls can be forwarded to other extension numbers(Local or Remote) and to local Voicemail Boxes if no one answers the call.
Field Type: Option buttons

Destination:
Field Type: [0-9]

Line Unavailable:

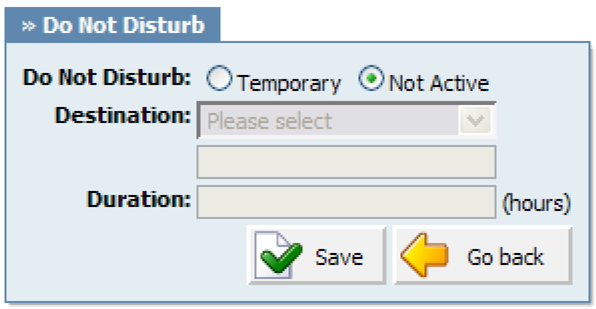
Forward all incoming calls if the line is unavailable.

Example: Calls can be forwarded to other extension numbers(Local or Remote) and to local Voicemail Boxes if line is, due to any reason, unavailable.
Field Type: Option buttons

Destination:
Field Type: [0-9]

20.2.4.5 Do Not Disturb

This service temporarily redirects all incoming calls to set destination number.
If you wish not to be disturbed set 'Destination'='Voicemail', Enter '1000' in the field bellow and set 'Duration'='1'. This will redirect all calls coming to your extension to extension 1000 voice box.



Do Not Disturb:

Activate the DND service

Example: This service can be set only on a temporary basis. Select 'Temporary' to activate it
Field Type: Option buttons

Destination:

Destination to be dialed once DND is enabled

Example: Select between 'Voicemail' or 'Call forward'. If Voicemail is set, then in the field bellow type the voice mailbox number, '1002' for example
Field Type: [0-9]

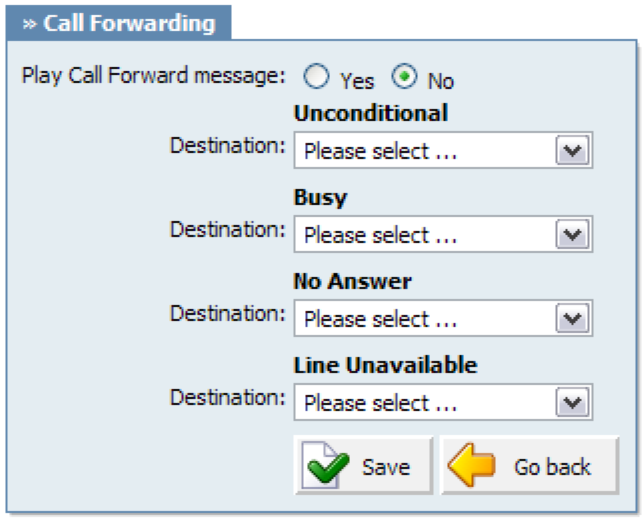
Duration:

Time in hours DND service will be active for

Example: Set '1' to enable the service for one hour
Field Type: [0-9]

20.2.4.6 Call Forwarding

This service forwards calls to other extensions depending on extension response/status.
Calls can be forwarded to other extensions/voice inboxes unconditionally, or only if extension is busy, nobody answers or when line is unavailable.



Play Call Forwarding Message:

Notify users of a transfer being made.

Example: If this option is set to 'Yes', the caller would hear a 'Please wait, your call is being forwarded. You're not being charged for the forwarding part of the call' message, notifying him that a transfer is being made.
Field Type: Option buttons

Unconditional:

Forward all incoming calls.

Example: Calls can be forwarded to other extension numbers(Local or Remote) and to local Voicemail Boxes.
Field Type: Option buttons

Destination:
Field Type: [0-9]

Busy:

Forward all incoming calls if the extension is busy.

Example: Calls can be forwarded to other extension numbers(Local or Remote) and to local Voicemail Boxes if the line has reached the maximum incoming calls limit.
Field Type: Option buttons

Destination:
Field Type: [0-9]

No Answer:

Forward all incoming calls if the extension doesn't answer the incoming call.

Example: Calls can be forwarded to other extension numbers(Local or Remote) and to local Voicemail Boxes if no one answers the call.
Field Type: Option buttons

Destination:
Field Type: [0-9]

Line Unavailable:

Forward all incoming calls if the line is unavailable.

Example: Calls can be forwarded to other extension numbers(Local or Remote) and to local Voicemail Boxes if line is, due to any reason, unavailable.
Field Type: Option buttons

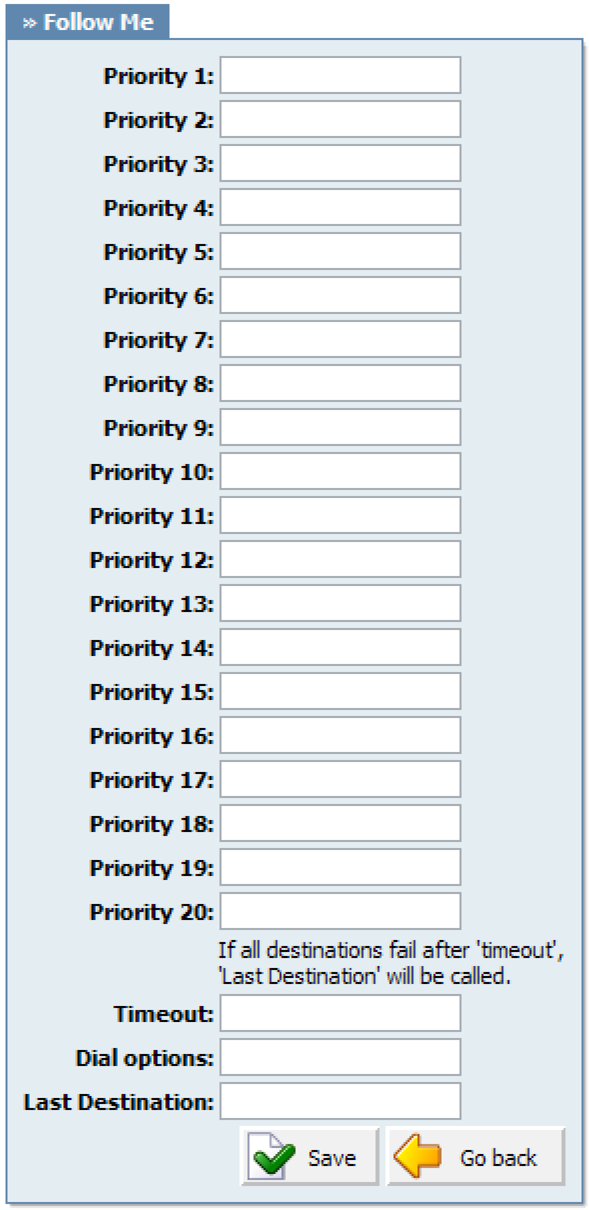
Destination:
Field Type: [0-9]

20.2.4.7 Follow Me

This service rings all provided destinations in a sequence. If call is not answered by any of the provided extensions, call gets transferred to 'Last Destination' extension.

For example, extension 5555 has the following extensions under 'Priority' fields: 1000, 1001, 1002 and cell phone 55510205. When someone calls extension 5555, extension 1000 will ring for 'Timeout' number of seconds. If noone answers extensions 1001 is dialled etc. If none of the Priority extensions answers the call, 'Last Destination' extension is called.

NOTE: If placing calls to mobile/proper number it may take 2-3 seconds until call is placed over Zaptel



Priority *:

Local/Proper/Mobile numbers to be dialed

Example: Enabling this option for extension 1005 and setting 'Priority 1' is set to '1008' and 'Priority 2' to '55510205' will dial local network number 1008. If noone answers during the 'Timeout' period local proper phone 55510205 is dialed etc...
Field Type: [0-9]

Timeout:

Ring time in seconds

Example: Time in seconds 'Priority' destinations will ring. If the call is not answered during this period, it gets transferred to next Priority number
Field Type: [0-9]

Dial Options:

Additional call properties

Example: This service can be assigned additional call properties, such as allowing the called party to transfer the call etc
Field Type: [a-z]

- **t** **-** Allow the called user to transfer the call by hitting #
- **T** **-** Allow the calling user to transfer the call by hitting #
- **r** **-** Generate a ringing tone for the calling party, passing no audio from the called channel(s) until one answers. Use with care and don't insert this by default into all your dial statements as you are killing call progress information for the user. Really, you almost certainly do not want to use this. Asterisk will generate ring tones automatically where it is appropriate to do so. 'r' makes it go the next step and additionally generate ring tones where it is probably not appropriate to do so.
- **R** **-** Indicate ringing to the calling party when the called party indicates ringing, pass no audio until answered. This is available only if you are using kapejod's bristuff.
- **m** **-** Provide Music on Hold to the calling party until the called channel answers. This is mutually exclusive with option 'r', obviously. Use m(class) to specify a class for the music on hold.
- **o** **-** Restore the Asterisk v1.0 Caller ID behaviour (send the original caller's ID) in Asterisk v1.2 (default: send this extension's number)
- **j** **-** Asterisk 1.2 and later: Jump to priority n+101 if all of the requested channels were busy (just like behaviour in Asterisk 1.0.x)
- **M(x)** **-** Executes the macro (x) upon connect of the call (i.e. when the called party answers)
- **h** **-** Allow the callee to hang up by dialing *
- **H** **-** Allow the caller to hang up by dialing *
- **C** **-** Reset the CDR (Call Detail Record) for this call. This is like using the NoCDR command
- **P(x)** **-** Use the Privacy Manager, using x as the database (x is optional)
- **g** **-** When the called party hangs up, exit to execute more commands in the current context.
- **G(context^exten^pri)** **-** If the call is answered, transfer both parties to the specified priority; however it seems the calling party is transferred to priority x, and the called party to priority x+1
- **A(x)** **-** Play an announcement (x.gsm) to the called party.
- **S(n)** **-** Hangup the call n seconds AFTER called party picks up.
- **d:** - This flag trumps the 'H' flag and intercepts any dtmf while waiting for the call to be answered and returns that value on the spot. This allows you to dial a 1-digit exit extension while waiting for the call to be answered - see also RetryDial
- **D(digits)** **-** After the called party answers, send digits as a DTMF stream, then connect the call to the originating channel.
- **L(x[:y][:z])** **-** Limit the call to 'x' ms, warning when 'y' ms are left, repeated every 'z' ms) Only 'x' is required, 'y' and 'z' are optional. The following special variables are optional for limit calls: (pasted from app_dial.c)
  - **+ LIMIT_PLAYAUDIO_CALLER** - yes|no (default yes) - Play sounds to the caller.
  - **+ LIMIT_PLAYAUDIO_CALLEE** - yes|no - Play sounds to the callee.
  - **+ LIMIT_TIMEOUT_FILE** - File to play when time is up.

- **+LIMIT_CONNECT_FILE** - File to play when call begins.
- **+LIMIT_WARNING_FILE** - File to play as warning if 'y' is defined. If LIMIT_WARNING_FILE is not defined, then the default behaviour is to announce ('You have [XX minutes] YY seconds').

- **f** **-** forces callerid to be set as the extension of the line making/redirecting the outgoing call. For example, some PSTNs don't allow callerids from other extensions than the ones that are assigned to you.
- **w** **-** Allow the called user to start recording after pressing *1 or what defined in features.conf, requires Set(DYNAMIC_FEATURES=automon)
- **W** **-** Allow the calling user to start recording after pressing *1 or what defined in features.conf, requires Set(DYNAMIC_FEATURES=automon)

Last Destination:

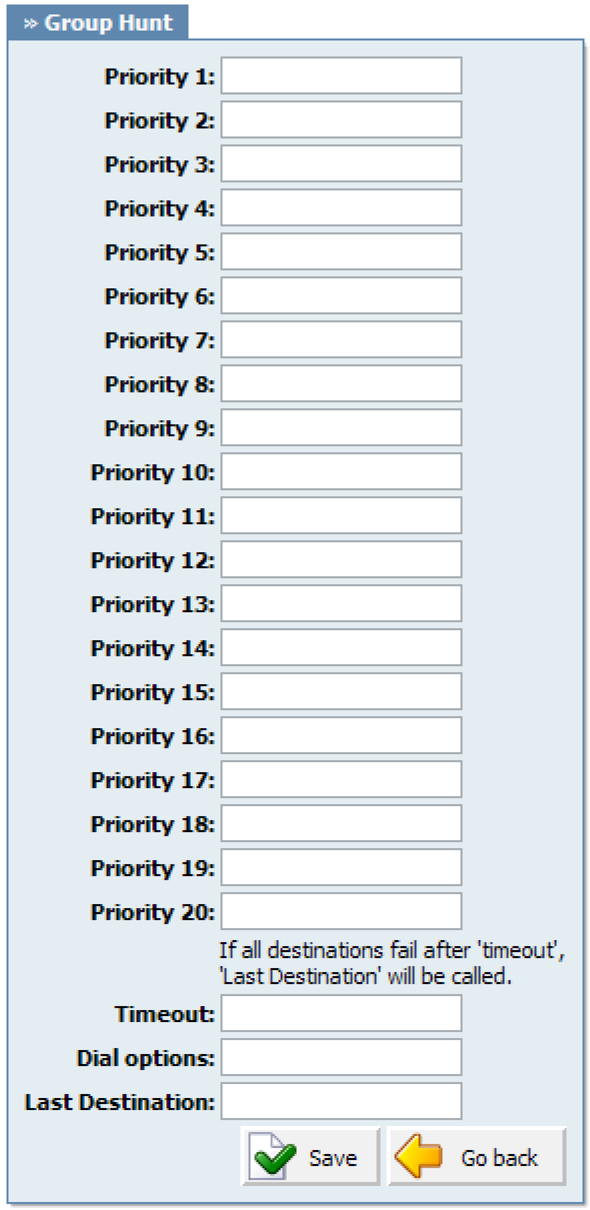
The last destination number dialed if none of the 'Priority' numbers answers the call

Example: Set this field to 1005. If none of the Priority extensions answers, 1005 is dialed
Field Type: [0-9]

20.2.4.8 Group Hunt

This service will dial all 'Priority *' provided extensions at the same time. If the call is not answered in the 'Timeout' period, by any of the 'Priority' extensions, the call is transferred to the 'Last Destination' extension.

For example, extension 1000 has 'Priority 1' set to '1002', and 'Priority 2' set to '1003'. Once extension 1010 dials the 1000, extensions 1002 and 1003 will ring at the same time(Note that dialed extension 1000 will not ring. In order to ring the dialed extension, simply provide its network number(1000) in one of the 'Priority' fields).



Priority *:

PBXware/Proper/Mobile numbers to be dialed

Example: Enabling this option for extension 1005 and setting 'Priority 1' is set to '1008' and 'Priority 2' to '55510205' will dial local network number 1008 and local proper phone 55510205 at the same time
Field Type: [0-9]

Timeout:

Ring time in seconds

Example: Time in seconds 'Priority' destinations will ring. If the call is not answered during this period, it gets transferred to the 'Last Destination'.
Field Type: [0-9]

Dial Options:

Additional call properties

Example: This service can be assigned additional call properties, such as allowing the called party to transfer the call etc
Field Type: [a-z]

- **t** **-** Allow the called user to transfer the call by hitting #
- **T** **-** Allow the calling user to transfer the call by hitting #
- **r** **-** Generate a ringing tone for the calling party, passing no audio from the called channel(s) until one answers. Use with care and don't insert this by default into all your dial statements as you are killing call progress information for the user. Really, you almost certainly do not want to use this. Asterisk will generate ring tones automatically where it is appropriate to do so. 'r' makes it go the next step and additionally generate ring tones where it is probably not appropriate to do so.
- **R** **-** Indicate ringing to the calling party when the called party indicates ringing, pass no audio until answered. This is available only if you are using kapejod's bristuff.
- **m** **-** Provide Music on Hold to the calling party until the called channel answers. This is mutually exclusive with option 'r', obviously. Use m(class) to specify a class for the music on hold.
- **o** **-** Restore the Asterisk v1.0 Caller ID behaviour (send the original caller's ID) in Asterisk v1.2 (default: send this extension's number)
- **j** **-** Asterisk 1.2 and later: Jump to priority n+101 if all of the requested channels were busy (just like behaviour in Asterisk 1.0.x)
- **M(x)** **-** Executes the macro (x) upon connect of the call (i.e. when the called party answers)
- **h** **-** Allow the callee to hang up by dialing *
- **H** **-** Allow the caller to hang up by dialing *
- **C** **-** Reset the CDR (Call Detail Record) for this call. This is like using the NoCDR command
- **P(x)** **-** Use the Privacy Manager, using x as the database (x is optional)
- **g** **-** When the called party hangs up, exit to execute more commands in the current context.
- **G(context^exten^pri)** **-** If the call is answered, transfer both parties to the specified priority; however it seems the calling party is transferred to priority x, and the called party to priority x+1
- **A(x)** **-** Play an announcement (x.gsm) to the called party.
- **S(n)** **-** Hangup the call n seconds AFTER called party picks up.
- **d:** - This flag trumps the 'H' flag and intercepts any dtmf while waiting for the call to be answered and returns that value on the spot. This allows you to dial a 1-digit exit extension while waiting for the call to be answered - see also RetryDial
- **D(digits)** **-** After the called party answers, send digits as a DTMF stream, then connect the call to the originating channel.
- **L(x[:y][:z])** **-** Limit the call to 'x' ms, warning when 'y' ms are left, repeated every 'z' ms) Only 'x' is required, 'y' and 'z' are optional. The following special variables are optional for limit calls: (pasted from app_dial.c)
  - **+ LIMIT_PLAYAUDIO_CALLER** - yes|no (default yes) - Play sounds to the caller.
  - **+ LIMIT_PLAYAUDIO_CALLEE** - yes|no - Play sounds to the callee.
  - **+LIMIT_TIMEOUT_FILE** - File to play when time is up.
  - **+LIMIT_CONNECT_FILE** - File to play when call begins.
  - **+LIMIT_WARNING_FILE** - File to play as warning if 'y' is defined. If

LIMIT_WARNING_FILE is not defined, then the default behaviour is to announce ('You have [XX minutes] YY seconds').

- **f** - forces callerid to be set as the extension of the line making/redirecting the outgoing call. For example, some PSTNs don't allow callerids from other extensions than the ones that are assigned to you.
- **w** - Allow the called user to start recording after pressing *1 or what defined in features.conf, requires Set(DYNAMIC_FEATURES=automon)
- **W** - Allow the calling user to start recording after pressing *1 or what defined in features.conf, requires Set(DYNAMIC_FEATURES=automon)

Last Destination:

The last destination number dialed if none of the 'Priority' numbers answers the call

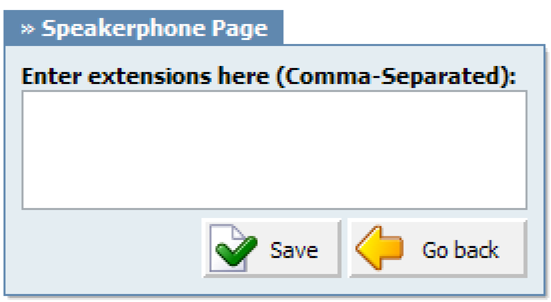
Example: Set this field to 1005. If none of the Priority extensions answers, 1005 is dialed
Field Type: [0-9]

20.2.4.9 Speakerphone Page

This service enables the message transmit to multiple phones by dialing '*399'.

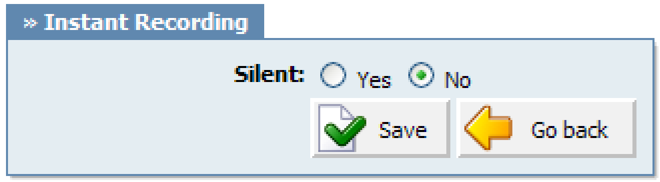
For example, set this field to '1000,1001,1002' for example. Now dial*399. Extensions 1000,1001,1002 will be paged. If UAD/Phone supports it, the call will automatically go to the speakerphone, otherwise it will just ring

NOTE: Phones by default have 10 seconds to auto answer.



20.2.4.10 Instant Recording

This service enables instant call recording, started anytime during the conversation, by dialing *159. For example, you may listen to any call made by extension 1000 for example. Simply dial *159 + 1000



Silent:

Should parties in conversation be informed that calls are being recorded

Example: With active 'Instant Recording' service, dial *159 anytime during the active call. From the point when you dial this code, the call will be recorded.
Field Type: Option buttons

20.2.4.11 Delete Recordings

Delete Recordings:

This service enables user to delete recorded calls via Self Care: CDR. For example, with this option enabled, user logs into Self Care, navigates to 'CDR', selects recorded message and clicks on 'Advanced: Delete Recordings'

20.2.4.12 Listen To Recordings

Listen To Recordings:

This service enables user to listed recorded calls via Self Care: CDR. For example, with this option enabled, user logs into Self Care, navigates to 'CDR', selects recorded message and clicks on 'Listen' button. Selected sound file will be downloaded to local computer from where it can be played in preferred audio player

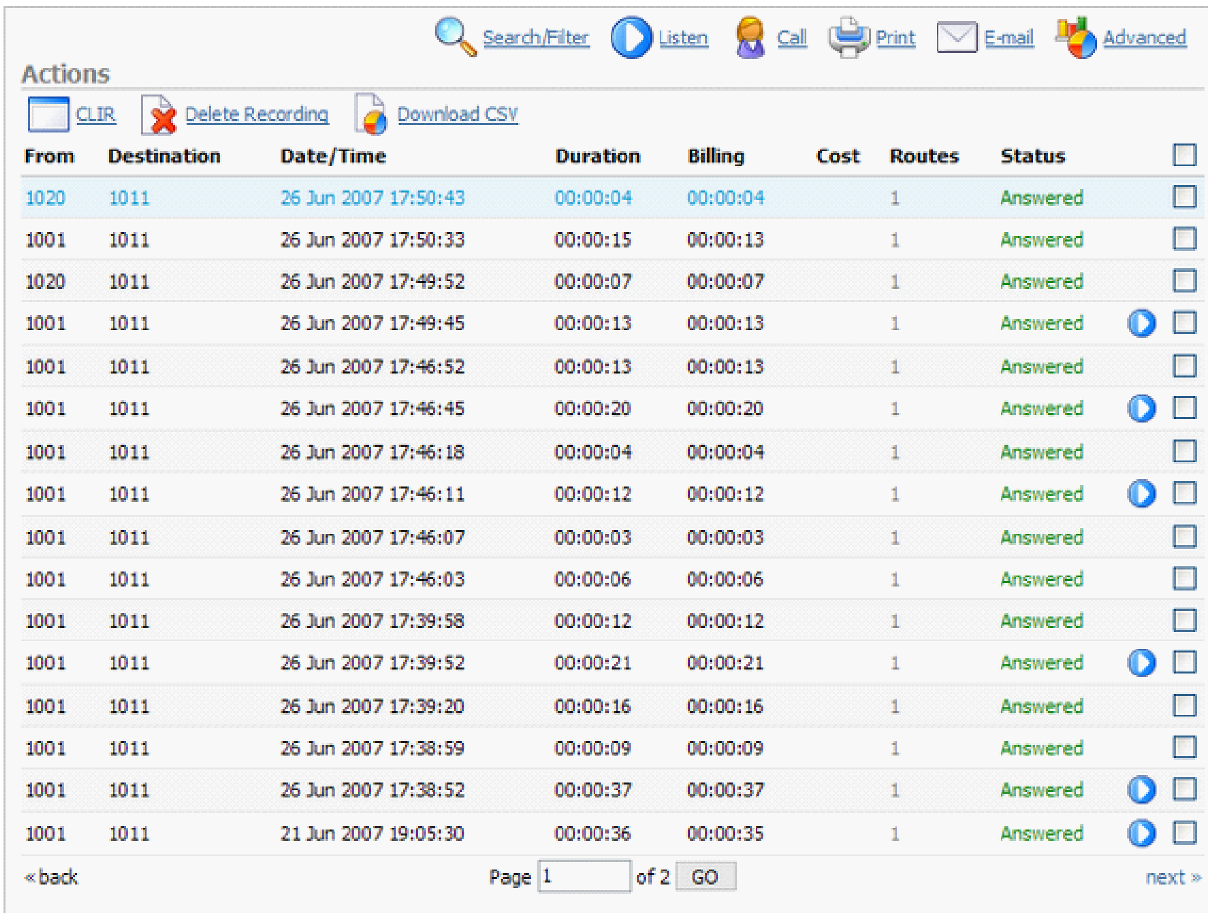
20.2.4.13 Remote Access

Remote Access:

This service enables user access system from remote location. For example, IVR '1001' has the following options set, '4'='Remote access: Destination'. Once remote user enters the IVR he will press '4', type in his extension number and PIN number (both confirmed with '#') and dial any local, mobile or proper number.

20.2.5 CDR

CDR (Call Detail Records) for all placed or received calls on the system. In addition to normal operation an authorized user is able to perform additional actions such as extensive search, listen to recorded calls, call any destinations listed and access advanced features.


Search/Filter Listen Call Print E-mail Advanced

Actions

CLIR Delete Recording Download CSV

From	Destination	Date/Time	Duration	Billing	Cost	Routes	Status
1020	1011	26 Jun 2007 17:50:43	00:00:04	00:00:04		1	Answered
1001	1011	26 Jun 2007 17:50:33	00:00:15	00:00:13		1	Answered
1020	1011	26 Jun 2007 17:49:52	00:00:07	00:00:07		1	Answered
1001	1011	26 Jun 2007 17:49:45	00:00:13	00:00:13		1	Answered
1001	1011	26 Jun 2007 17:46:52	00:00:13	00:00:13		1	Answered
1001	1011	26 Jun 2007 17:46:45	00:00:20	00:00:20		1	Answered
1001	1011	26 Jun 2007 17:46:18	00:00:04	00:00:04		1	Answered
1001	1011	26 Jun 2007 17:46:11	00:00:12	00:00:12		1	Answered
1001	1011	26 Jun 2007 17:46:07	00:00:03	00:00:03		1	Answered
1001	1011	26 Jun 2007 17:46:03	00:00:06	00:00:06		1	Answered
1001	1011	26 Jun 2007 17:39:58	00:00:12	00:00:12		1	Answered
1001	1011	26 Jun 2007 17:39:52	00:00:21	00:00:21		1	Answered
1001	1011	26 Jun 2007 17:39:20	00:00:16	00:00:16		1	Answered
1001	1011	26 Jun 2007 17:38:59	00:00:09	00:00:09		1	Answered
1001	1011	26 Jun 2007 17:38:52	00:00:37	00:00:37		1	Answered
1001	1011	21 Jun 2007 19:05:30	00:00:36	00:00:35		1	Answered

« back Page 1 of 2 GO next »

From:
Extension number the call was made from
Example: If call was made from extension 1001 to extension 1004, '1001' is displayed here.
Field Type: Display

Destination:
Extension number the call was made to
Example: If call was made from extension 1001 to extension 1004, '1004' is displayed here.
Field Type: Display

Date/Time:
Date and Time when the call was made
Example: 04 Oct 2006 10:44:10
Field Type: Display

Duration:
Call duration time in hh:mm:ss format
Example: 00:12:45
Field Type: Display

Billing:
Time billed by the system
Example: 00:12:45
Field Type: Display

Cost:
Total cost of the call calculated through a service plan
Example: 0.71
Field Type: Display

Routes:
Number of system routes
Example: If call goes directly from extension to destination - '1' will be displayed here. If call has entered the IVR and was then redirected to Queue and then Agent answers the call '2' will be displayed here (IVR and Queue were used)
Field Type: Display

Status:
Displays the call status
Example: Depending on whether a call was answered or not, this field value may have the following content:

- Answered
- Not Answered
- Busy
- Error

Field Type: Display


This icon is displayed once a call is recorded and 'Delete' or 'Listen' enhanced service is active


This is a box used with the CDR commands to select a desired call

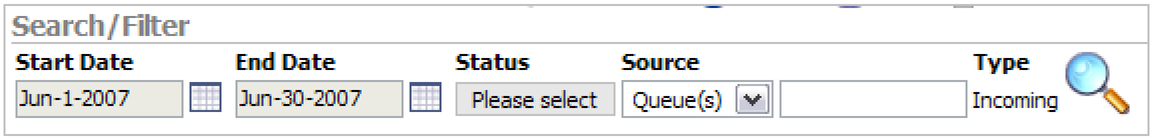
NOTE: Disk Space Used By Call Recording:
With continuously tone 60 seconds:

- wav49 = 84.5kb
- wav = 833.0kb
- gsm = 85.0kb

With continuously silent tone (without sound) 60 seconds:

- wav49 = 84.0kb
- wav = 827.0kb
- gsm = 84.0kb

20.2.5.1 Search/Filter



Start Date:

Select a Search/Filter start date

Example: Click on a small 'Calendar' icon next to a field and select desired date
Field Type: Option button

End Date:

Select a Search/Filter end date

Example: Click on a small 'Calendar' icon next to a field and select desired date
Field Type: Option button

Status:

Search calls by selecting desired call status

Example: Click on a 'Please Select' button and select one of the available fields:
- All
- Answered
- Not Answered
- Busy
- Error

Field Type: Select box

Type:

Search calls based on the type of calls

Example: Click the 'Type' button and select one of the available fields:
- All
- Outgoing
- Incoming

Field Type: Select box

NOTE: After making any changes to search filter, be sure to click the  search icon

20.2.5.2 Actions

In this chapter we will cover:

20.2.5.2.1 Listen

Listen:

Once the 'Listen'  icon is displayed next to a call record it means that the specific call was recorded.

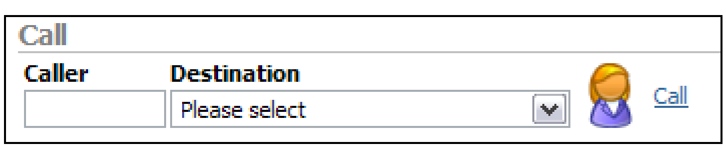
Example: To play recorded calls, check the box next to a 'Listen' icon and click 'Listen'. Browser will prompt you to open the sound file in your favorite audio player or to download the sound file.
Field Type: Option button

NOTE: By default the sound format is available as a .gsm file. To change the recording format go to: 'Settings: Servers: Edit: Recordings format' and select one of the available sound formats:

- gsm
- wav
- wav49 and
- ogg

20.2.5.2.2 Call

To establish a call between any PBXware extension with a listed extensions you have to provide only two things. The Caller $EXTENSION number and the $DESTINATION extension



Caller

PBXware extension that will make a call

Example: Provide any PBXware extension number here, 1001 for example
Field Type: [0-9]

Destination:

Destination extension that will be dialed by 'Caller' extension

Example: To select a destination extension, first check a box next to a CDR record. This field will display two extensions listed under 'From' and 'Destination' selected record
Field Type: Select button

NOTE: After setting 'Caller' and 'Destination' extensions click the  call icon

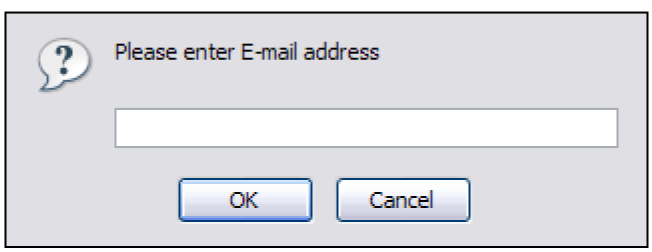
20.2.5.2.3 Print

Check the box next to a call record and click the 'Print' button. This action will open a new popup window with the printing interface.

From	Destination	Date/Time	Duration	Status
5555	4000	30 Jan 2007 11:25:02	00:00:24	Answered
1000	4000	29 Jan 2007 17:21:20	00:00:14	Answered
1000	4000	29 Jan 2007 17:20:45	00:00:25	Answered
1000	4000	29 Jan 2007 17:03:13	00:00:27	Answered
1000	4000	29 Jan 2007 17:02:47	00:00:14	Answered
1000	4000	29 Jan 2007 16:59:52	00:00:24	Answered
1000	4000	29 Jan 2007 16:59:50	00:00:09	Answered
1000	4000	29 Jan 2007 16:59:49	00:00:00	Answered
1000	4000	29 Jan 2007 16:59:29	00:00:19	Answered
1000	4000	29 Jan 2007 16:59:20	00:00:27	Answered
1000	4000	29 Jan 2007 16:58:48	00:00:38	Answered
1000	4000	29 Jan 2007 16:58:45	00:00:33	Answered
1000	4000	29 Jan 2007 16:58:33	00:00:13	Answered
1000	4000	29 Jan 2007 16:58:22	00:00:21	Answered
1000	4000	29 Jan 2007 16:58:12	00:00:15	Answered
1000	4000	29 Jan 2007 16:57:50	00:00:19	Answered

20.2.5.2.4 Email

Check the box next to a call record and click the 'Email' button. A small popup dialog will appear. Provide email address here and click 'OK' button to send the records.



20.2.6 Settings

These options mimic the functions of an answering machine but with many additional features added. Voice messages are saved on central file-system location instead on a UAD/Phone.

Accessing voice-box:

To access voice-box dial '*123', enter extension PIN and follow the instructions.

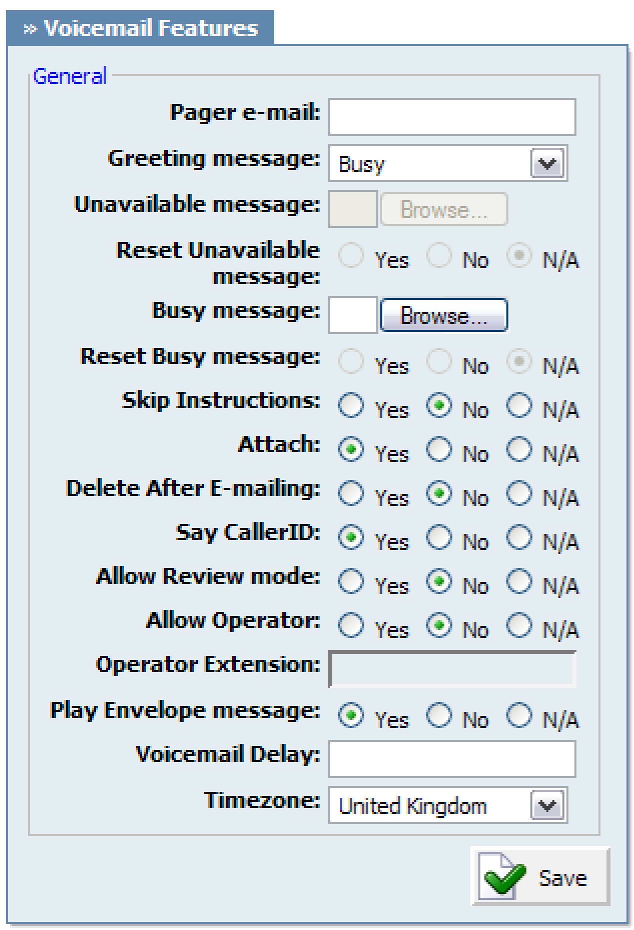
Leaving a voice message:

When user is transferred to extension's voice-box, 'Please leave a detail message after the tone. If you would like to speak to the operator, press 0' message will be heard.

Two options are available:
1. Leave a voice message(ended by pressing '#' key or hanging up), or
2. Reach an operator by dialing '0'

If '0' is dialed, 'Press 1 to accept this recording, otherwise please continue to hold' message will be heard.

Two options are available:
1. Press '1' to save your message, after which the operator will be dialed. 'Please hold while i try that extension' message will be heard, or
2. Continue to hold, which will delete any left messages, after which the operator will be dialed. 'Message deleted, please hold while i try that extension' message will be heard.



Pager e-mail:

Pager e-mail address associated with the voice box.

Example: When A calls B and leaves a voice message, B will get a pager email notification about new voice message.
Field Type: [a-z] [0-9] [@._-]

Greeting message:

Greeting message played to users upon entering the voice box.

Example: When A gets to B's voice box, the selected 'Greeting message' is played to A before he is allowed to leave a message.
Field Type: Select box

Unavailable message:

Upload unavailable message

Example: Click on the 'Browse' button and select a sound file from local computer to upload it as a custom unavailable message.
Field Type: Button

Reset Unavailable message:

Resets the user recorded/uploaded unavailable message.

Example: Custom unavailable messages can be recorded through UAD/Phone or uploaded to voice box through Self Care. To revert to default system unavailable message select 'Yes' and save the extension settings.
Field Type: Option buttons

Busy message:

Upload busy message

Example: Click on the 'Browse' button and select a sound file from local computer to upload it as a custom busy message.
Field Type: Button

Reset Busy message:

Resets the user recorded/uploaded busy message.

Example: Custom busy messages can be recorded through UAD/Phone or uploaded to voice box through Self Care. To revert to default system busy message select 'Yes' and save the extension settings.
Field Type: Option buttons

Skip Instructions:

Skip the instructions on how to leave a voice message.

Example: Once user A reaches the dialed voice box, if this option is set to 'Yes', A will hear the 'Greeting message', and then be transferred directly to the 'beep' sound.
Field Type: Option buttons

Attach:

Send the voice message as an attachment to user email.

Example: Once B gets the new voice message, if this option is set to 'Yes', the message sound file will be attached to the new voicemail notification email.
Field Type: Option buttons

Delete After E-mailing:

Delete voice message after sending it as an attachment to user email.

Example: Once B gets the new voice message, if this option is set to 'Yes', the message will be deleted from the voice box after it has been emailed to B.
Field Type: Option buttons

Say Caller ID:

Announce the extension number from which the voice message has been recorded.

Example: If this option is set to 'Yes', when checking voicemail, 'From phone number {$NUMBER}' message will be heard.
Field Type: Option buttons

Allow Review mode:

Allow B to review the voice message before committing it permanently to A's voice box.

Example: B leaves a message on A's voice box, but instead of hanging up, he presses '#'. Three options are offered to B:

- Press 1 to accept this recording
- Press 2 to listen to it
- Press 3 to re-record your message

Field Type: Option buttons

Allow Operator:

Allow B to reach an operator from within the voice box.

Example: B leaves a message on A's voice box, but instead of hanging up, B presses '#'.
- 'Press 0 to reach an operator' message played (Once '0' is pressed, user is offered the following options):
  - Press 1 to accept this recording (If selected, 'Your message has been saved. Please hold while I try that extension' is played and operator is dialed)
  - Or continue to hold (If B holds for a moment, 'Message deleted. Please hold while I try that extension' is played and operator is dialed)

Field Type: Option buttons

Operator Extension:

Local extension number that acts as an operator.

Example: If A's voice box has an option 'Allow Operator' set to 'Yes', all users dialing '#0' inside the voice box will reach this operator extension.
Field Type: [0-9]

Play Envelope message:

Announces the Date/Time and the Extension number from which the message was recorded.

Example: Once voice box is checked for new messages, if this option is set to 'Yes', 'Received at {$DATE}. From phone number {$NUMBER}' will be played, giving more details about the message originator.
Field Type: Option buttons

Voicemail Delay:

How long to pause in seconds, before asking user for PIN/Password.

Example: Some UADs/Phones have tendency to garble the beginning of a sound file. Therefore, user checking the voice box, when asked for password would hear '...sword' instead of 'Password'. Setting this field to 1-2 seconds will provide long enough gap to fix this anomaly.
Field Type: [0-9]

Timezone:

Sets the correct date/time stamp.

Example: By setting the correct time zone, user would always be notified of the exact date/time voice message was left on their box. Set the correct time zone if user is located in different time zone then

PBXware.
Field Type: Select box

NOTE: Timezones are taken from '/usr/share/zoneinfo' system directory

NOTE: Disk Space Used By Voicemail Recording
With continuously tone 60 seconds:
- wav49 = 91.0kb
- wav = 863.0kb
- gsm = 91.0kb

With continuously silent tone (without sound) 60 sec:
- wav49 = 0.38kb
- wav = 3.0kb
- gsm. = 0.32k b

20.3 Help

A click on 'Help' button opens PBXware online support files

20.4 Logout

Logs out user from Self Care

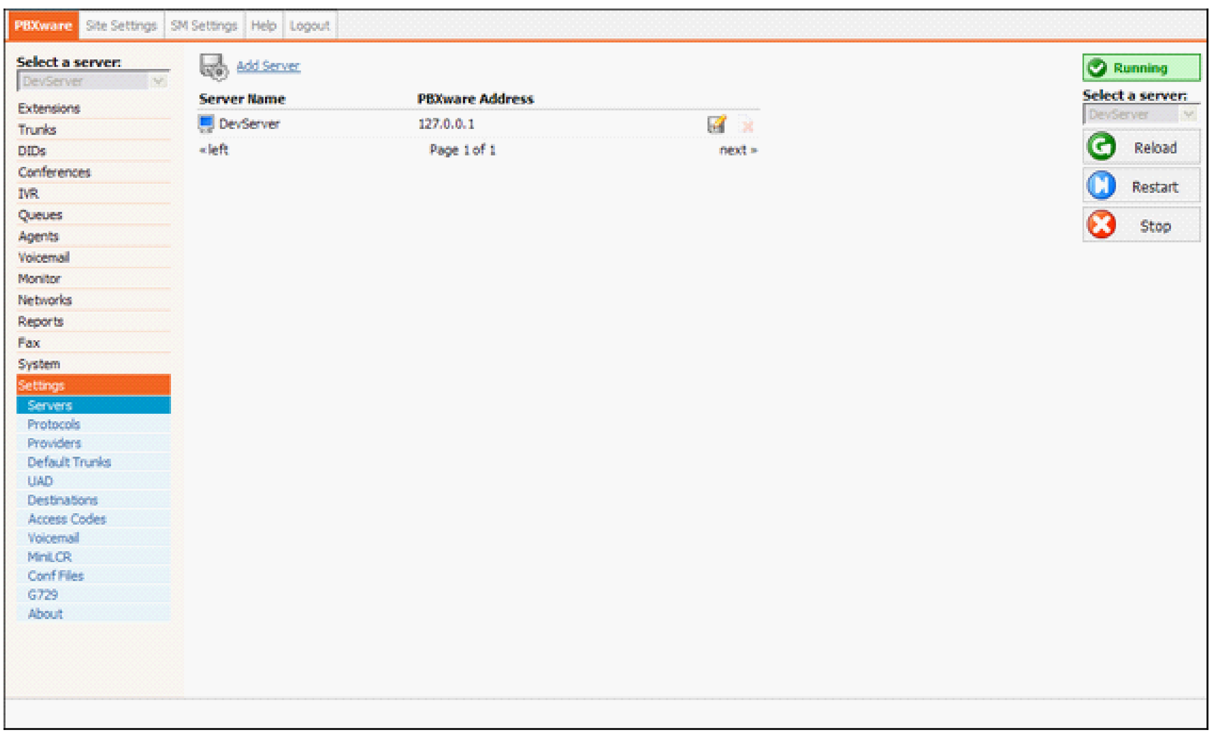
Part XXI

21 Settings

Settings are used to set global system variables. These settings are applied on entire system if not otherwise specified on lower level.

For example: 'Settings: Voicemail: Send Attachment'='Yes' will set 'Set Attachment' for all local and remote extensions. But each extension can override this rule by setting 'Extensions: Edit: Advanced Options: Attach'='No'

In this chapter we will cover:

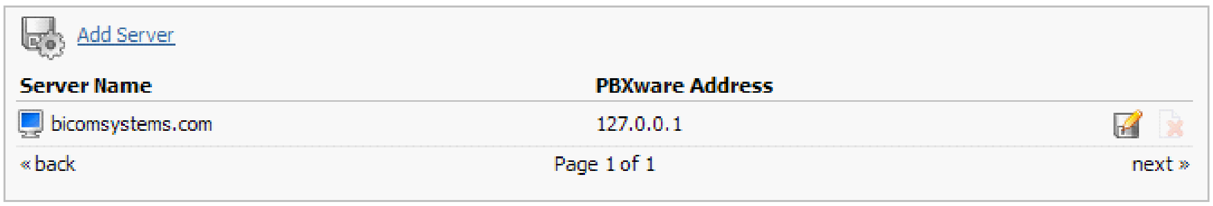


21.1 Servers

PBXware system administration allows for an unlimited number of servers to be administered from a single administration interface. The system servers can in following modes:

Local: This is single system server providing services to one organization at one location just as any other PBXware.

Network: When system servers are in network mode, all servers share common dial plan. A single dial plan allows any organization to have a network of servers located anywhere in the world.



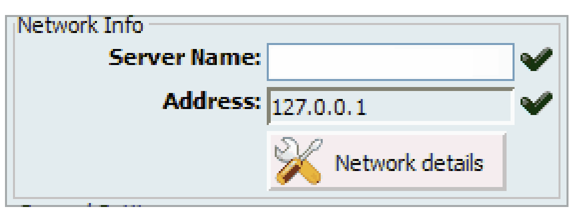
Server Name:
Unique, custom set, server name on the network
Example: system@domain.com
Field Type: Display

PBXware Address:
System public IP address
Example: 127.0.0.1
Field Type: Display

Edits the server configuration
Example: Click to edit server configuration
Field Type: Button

Deletes a server from the system
Example: Click to delete a server from the system
Field Type: Button

21.1.1 Network Info

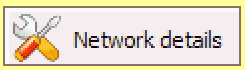


Server Name:

Custom server name(occasionally used in system notifications)

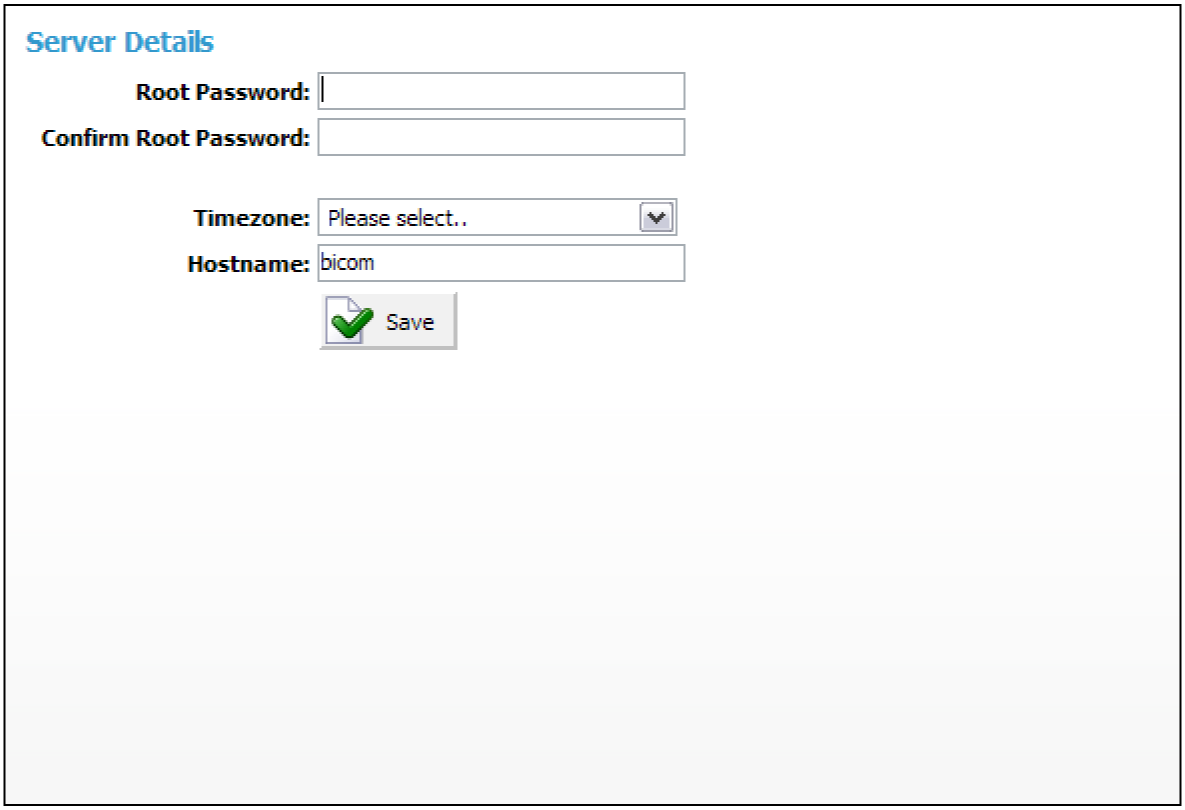
Example: domain.com
Field Type: [a-z][0-9]

Address:
Default IP/Hostname of local PBXware server. This value can be set for remote PBXware server as well if the whole network is configured for multi server operation
Example: 127.0.0.1
Field Type: Display, IP Address



Network details configuration window
Example: Click to edit network configuration
Field Type: Button

21.1.1.1 Server Details



Root Password:

Set the root password

Example: PBXware prompts for this password during the system/ssh login and when accessing system services throughinterface
Field Type: [a-z][0-9]

Confirm Password:

Confirm the root password

Example: Re-type the Root Password entered in the field above
Field Type: [a-z][0-9]

Time zone:

Time zone PBXware is located at

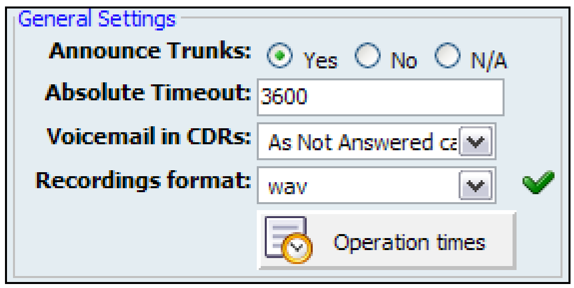
Example: Select the appropriate time zone, for example 'USA/East-coast
Field Type: Select box

Hostname:

The name given to machine which will identify the system on the network

Example: hostname
Field Type: [a-z][0-9]

21.1.2 General Settings



Announce Trunks:

Announce over which trunk call goes through

Example: John dials 55510205 and this call goes over secondary default system trunk. John will hear 'Using secondary trunk to terminate your call' message.
Field Type: Option buttons

Absolute Timeout:

Maximum time a call can last (in seconds)

Example: if '3600' is set in this field, that will make all calls end after 1 hour (1h = 3600 seconds)
Field Type: [0-9]

Voicemail in CDRs:

Sets how the calls that were unanswered and redirected to voicemail are displayed in CDR

Example: A call was made to extension 1000 but was not answered. The caller gets redirected to voicemail. If 'As Not Answered call' is set, CDR will display this voicemail redirection as 'Unanswered' call. If no option is set same call will be displayed as 'Answered'
Field Type: [0-9]

Recordings format:

Format used for saving the system call and voicemail recordings. You can read more details about disk space usage on the bottom of this chapter

Example: Choose one of the following formats: gsm, wav, wav49 and ogg. If wav is selected, all call recordings and voicemail recordings will be save in this format.
Field Type: Select box

Operation Times:

System operation time

Example: For more explanation click here
Field Type: Option buttons

NOTE: Disk Space Used By Call Recording:

With continuously tone 60 seconds:

- wav49 = 84.5kb
- wav = 833.0kb
- gsm = 85.0kb

With continuously silent tone (without sound) 60 seconds:

- wav49 = 84.0kb
- wav = 827.0kb
- gsm = 84.0kb

NOTE: Disk Space Used By Voicemail Recording
With continuously tone 60 seconds:

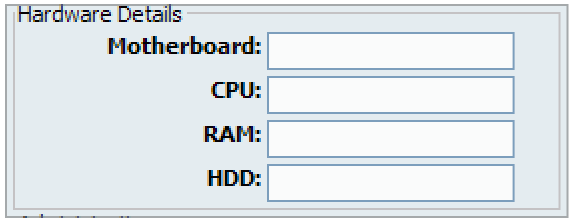
- wav49 = 91.0kb
- wav = 863.0kb
- gsm = 91.0kb

With continuously silent tone (without sound) 60 sec:

- wav49 = 0.38kb
- wav = 3.0kb
- gsm. = 0.32k b

21.1.3 Hardware Details

Server hardware specifications are set here.



Motherboard:
Motherboard info
Example: ASUS A8N-SLI Deluxe Motherboard
Field Type: [a-z][0-9]

CPU:
CPU info
Example: Intel Pentium 4 3 GHz
Field Type: [a-z][0-9]

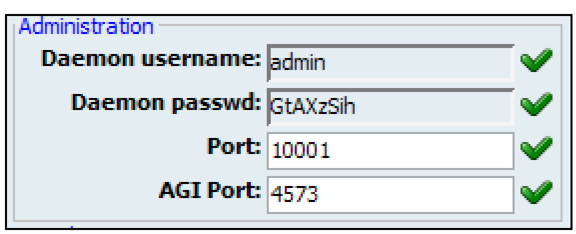
RAM:
RAM info
Example: Kingston KVR266X64C25/512 SDRAM
Field Type: [a-z][0-9]

HDD:
HDD info
Example: Western Digital Caviar 80 GB
Field Type: [a-z][0-9]

21.1.4 Administration

This section enables remote administration of the system.

NOTE: Daemon username and password must be the same as set in: '/home/servers/pbxware/pw/pbxware/etc/pbxware.ini' file



Admin username:

Administrator username

Example: admin
Field Type: [a-z][0-9]

Admin passwd:

Administrator password

Example: admin
Field Type: [a-z][0-9]

Port:

PBXware connection port

Example: 10001
Field Type: [0-9]

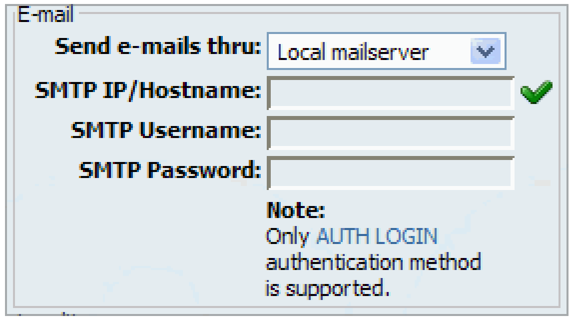
AGI Port:

Agi connection port

Example: 4573
Field Type: [0-9]

21.1.5 Email

System during its operations sends email notifications and alerts to various users and administrator. These email can be sent using built in 'local mail server' or remote SMTP server.



Send e-mails thru:

Select the method system delivers email messages through

Example: System can send email messages using the 'Local mailserver' or 'Remote SMTP server'. If using remote, please provide Hostmane, Username and Password details
Field Type: [0-9]

SMTP IP/Hostname:

SMTP IP Hostname of remote mail server. 'Send e-mails thru' must be set to 'Remote SMTP server'

Example: 192.168.1.1/mail.domain.com
Field Type: [a-z][0-9], IP address

SMTP Username:

Username for remote SMTP server. 'Send e-mails thru' must be set to 'Remote SMTP server'

Example: username
Field Type: [a-z][0-9]

SMTP Password:

Password for remote SMTP server. 'Send e-mails thru' must be set to 'Remote SMTP server'

Example: username
Field Type: [a-z][0-9]

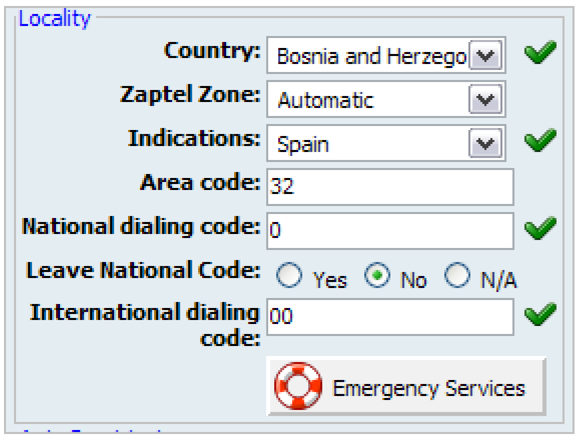
NOTE:

- Only AUTH LOGIN authentication method is supported. Please visit http://www.ietf.org/rfc/rfc2554.txt for more.

- To see the SMTP logs navigate to 'Reports: SMTP log'

21.1.6 Locality

Locality sets from where the system is operating from



Country:

Country PBXware is located at or operated from

Example: If PBXware is located in USA set USA here
Field Type: Select box

Zaptel Zone:

Overrides Automatic Country Detection

Example: It is recommended to keep this setting always set to 'Automatic'
Field Type: Select box

Indications:

Which indications (Ringing, Busy etc sounds) are to be used by the PBXware

Example: If the system is located in USA, set USA here, otherwise select the closest country to yours
Field Type: Select box

Area Code:

Area code the system is located or operating from

Example: If PBXware is located in New York, set New York area code here (212
Field Type: [0-9]

National Dialing Code:

Code needed for dialing national destinations

Example: 1(USA), 0(United Kingdom, Germany...)
Field Type: [0-9]

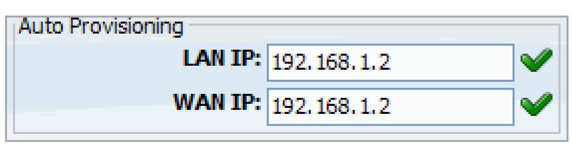
International Dialing Code:

Code needed for dialing international destinations

Example: 011(USA), 00(United Kingdom, Germany...)
Field Type: [0-9]

21.1.7 Auto Provisioning

Auto provisioning sets values to be used for the auto provisioning system in order to create auto provisioning files correctly depending on the UAD locality (local/remote).



LAN IP:

Local area network IP address used to auto provision local UADs

Example: 192.168.1.2
Field Type: IP address

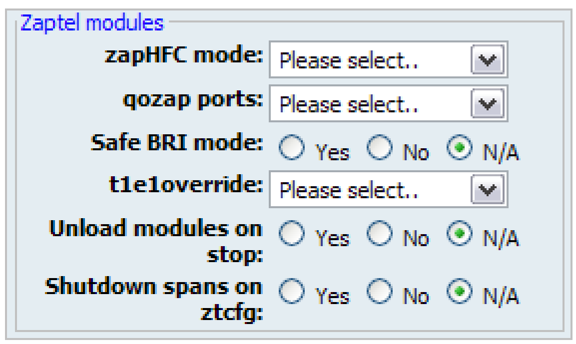
WAN IP:

Wide area network IP address used to auto provision remote UADs

Example: 192.168.1.2
Field Type: IP address

21.1.8 Zaptel Modules

NT is the abbreviation of 'Network Terminator'. TE is the abbreviation for 'Terminal Equipment'.



zapHFC mode:

Set BRI card mode

Example:
- TE mode
- NT mode
- TE, TE mode (2 cards)
- NT, TE mode (2 cards)
- TE, NT mode (2 cards)
- NT, NT mode (2 cards)

Field Type: Select box

qozap ports:

Set the QuadBRI card mode.

Example:
- quadBRI defaults
- TE, TE, TE, TE

- NT, TE, TE, TE
- TE, NT, TE, TE
- NT, NT, TE, TE
- TE, TE, NT, TE
- NT, TE, NT, TE
- TE, NT, NT, TE
- NT, NT, NT, TE
- TE, TE, TE, NT
- NT, TE, TE, NT
- TE, NT, TE, NT
- NT, NT, TE, NT
- TE, TE, NT, NT
- NT, TE, NT, NT
- TE, NT, NT, NT
- NT, NT, NT, NT

Field Type: Select box

Safe BRI mode

Should safe BRI mode be activated

Example: Select 'Yes' to enable
Field Type: Option button

t1e1override:

Overrides the jumper settings. The decimal value between 0 and 15 correspond to binary from 0000 to 1111. 0 is T1 and 1 is E1.

Example:
(Decimal | Binary)
0 | 0000
1 | 0001
2 | 0010
3 | 0011
4 | 0100
5 | 0101
6 | 0110
7 | 0111
8 | 1000
9 | 1001
10 | 1010
11 | 1011
12 | 1100
13 | 1101
14 | 1110
15 | 1111
Field Type: Select box

Unload modules on stop:

Should PBXware unload all modules used by ISDN cards when PBXware stops

Example: Select 'Yes' and Reboot the PBXware if ISDN card cannot be auto-configured
Field Type: Option button

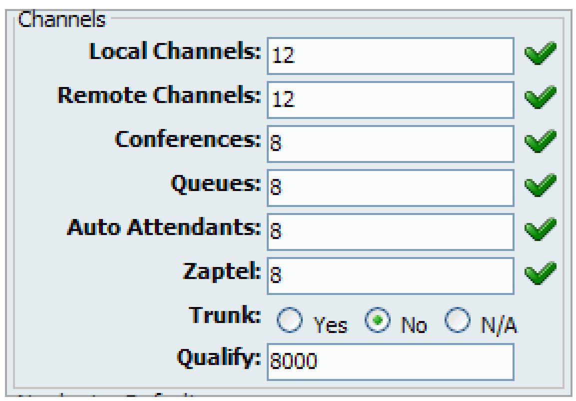
Shutdown spans on ztcfg:

Should card ports be shutdown when ztcfg command is executed

Example: This option is enabled if using Junghans card in combination with other cards
Field Type: Option button

21.1.9 Channels

Depending on CPU power of the server a custom number of channels can be assigned for various channel types



Local Channels:

Total number of all channels used by local UADs

Example: 12
Field Type: [0-9]

Remote Channels:

Total number of all channels used by remote UADs

Example: 12
Field Type: [0-9]

Conferences:

Total number of all system conferences

Example: 8
Field Type: [0-9]

Queues:

Total number of all system ACD queues

Example: 8
Field Type: [0-9]

Auto Attendants:

Total number of all system IVRs

Example: 8
Field Type: [0-9]

Zaptel:

Total number of all system trunks using ZAPTEL protocol

Example: 8
Field Type: [0-9]

Zaptel:

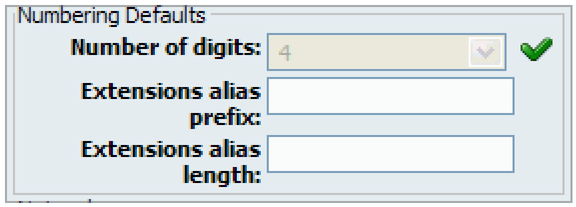
Total number of all system trunks using ZAPTEL protocol.

NOTE: The System will limit the number of channels in order to achieve and maintain excellent calls and other services' quality

Example: 8
Field Type: [0-9]

21.1.10 Numbering Defaults

Numbering default is set during the initial system set up in order to set how many digits the system will use as default. Available options are : 2, 3, and 4 digits.



Number of Digits:

Number of digits used by the system to create local extensions, IVRs, Queues, Voicemail boxes, Conferences etc.

Example: This option is available for settings only during setup wizard install process. In order to change number of digits after setup wizard, please remove all Extensions, DIDs, Conferences ... (all apps with network number). Recommended value for this field is 4
Field Type: Select box

Extensions alias prefix:

Digit assigned before dialled number on systems that have switched from 3 to 4 digit extensions for example

Example: If your old system have used 3 digits and you wish to switch to 4 for example, set '1' here to assign '1' as a prefix to all system extensions. This was, the old 300 extension becomes 1300.
Field Type: [0-9]

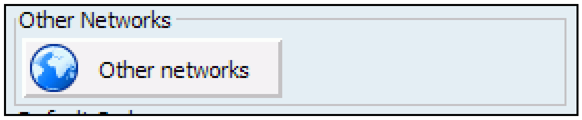
Extensions alias length:

Length of the old numbering system

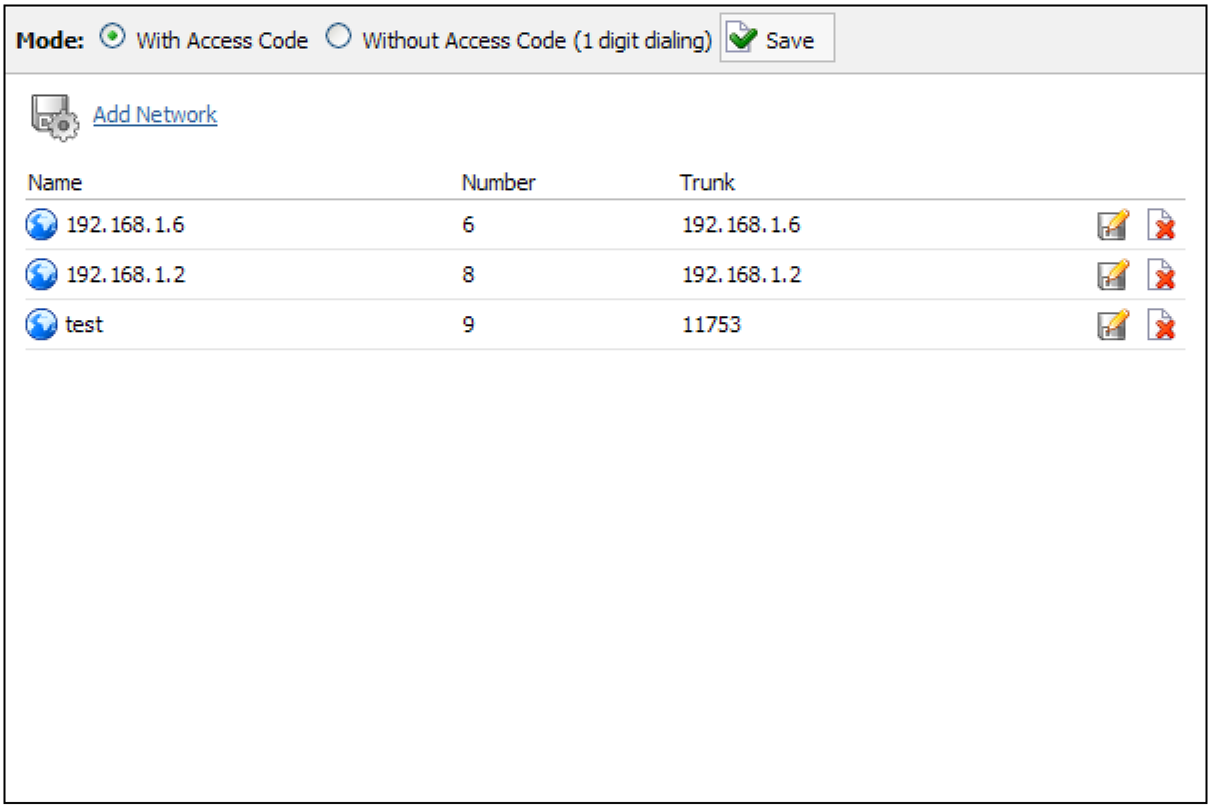
Example: If your old system have used 3 digits type 3 here
Field Type: [0-9]

21.1.11 Other Networks

The system can be part of the 'default' PBXware network where all extensions share same unified dial plan. This is achieved by selecting from select box which PBXware network does system belongs to.



A click on 'Other network' will open following options:



Mode:

Sets the way other PBXware network are dialed

Example:
- With Access Code - Access code + network number + extension (e.g. *188 8 1000)
- Without Access Code - network number only + extension (e.g. 8 1000)

Field Type: Option buttons

Name:
Other network name
Example: Network London
Field Type: Display

Number:

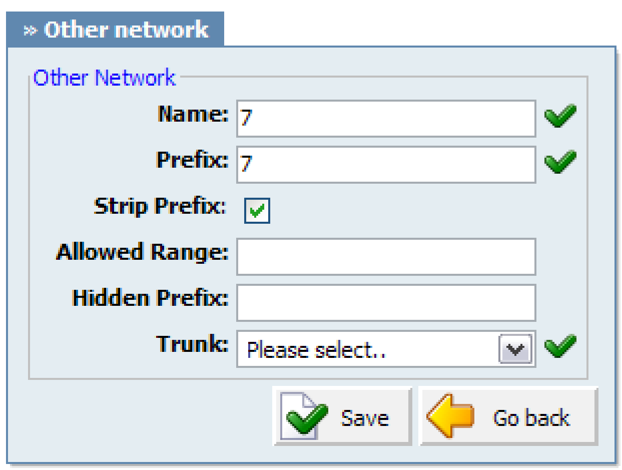
Other network access number
Example: 8
Field Type: Display

Trunk:
Trunk used once other network number is dialed
Example: 2554433
Field Type: Display


Edits the other network configuration
Example: Click to edit other network configuration
Field Type: Button


Deletes other network connection
Example: Click to delete other network connection from the system
Field Type: Button

21.1.11.1 Add/Edit Network



Name:

Custom Other Network name/identifier

Example: London FO/7
Field Type: [a-z][0-9]

Prefix:

Number used to access Other Network(Up to 3 digits allowed)

Example: If this field is set to '7', dial '*188 7 {NETWORK NUMBER}'
Field Type: [0-9]

Strip Prefix:

Should the 'Prefix' number be stripped once dialing Other network

Example: If 'Prefix' field is set to '7' and this field is enabled, once user dials *188 7 55510205 system will dial 55510205. If this field is disabled, 755510205 will be dialed.
Field Type: Option buttons

Allowed Range:

Number range allowed to be dialled after the Other Network Access Code

Example: If this field is set to '2', only extensions on Other Network starting with number 2 will be allowed to dial. If you dial *188 7 1002, our call will fail. But, if we dial *188 7 2000, our call will be transferred to extension 2000.
Field Type: [a-z][0-9]

Hidden Prefix:

Prefix number added before dialed number

Example: If this field is set to 212 and we dialed *188 7 55510205, 21255510205 will be dialed. This is useful is provider requires a certain code before dialing the destination, area code inserted automatically etc...
Field Type: [a-z][0-9]

Trunk:

Select trunk that will be used once 'Number' is dialed

Example: London FO Trunk
Field Type: Select box

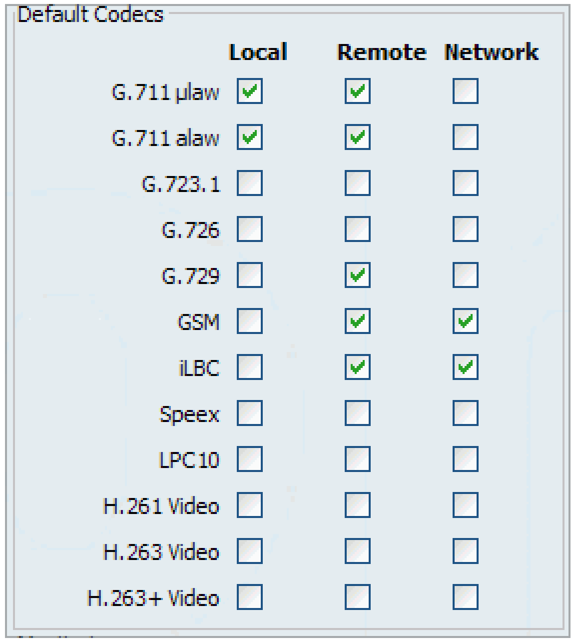
21.1.12 Default Codecs

Default codecs can be set for the following groups:

Local - Local extensions

Remote - Remote extensions and Trunks
Network - PBXware network(two or more servers)

NOTE: Once a local/remote extension or a network is added/edited, only the codecs allowed here will be available for the extension/network usage.



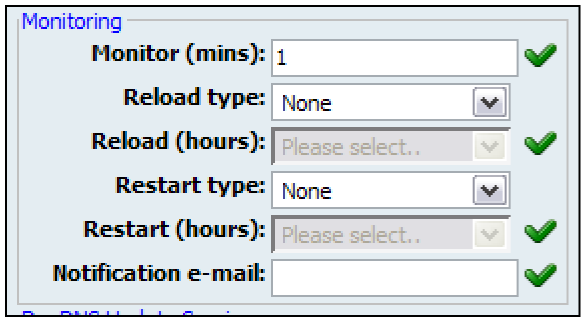
Available Codecs:

- **ITU G.711 ulaw** - 64 Kbps, sample-based, used in US
- **ITU G.711 alaw** - 64 Kbps, sample-based, used in Europe
- **ITU G.723.1** - (5.3/6.3 Kbps, 30ms frame size
- **ITU G.726** - 16/24/32/40 Kbps
- **ITU G.729** - 8 Kbps, 10ms frame size
- **GSM** - 13 Kbps (full rate), 20ms frame size
- **iLBC** - 15Kbps,20ms frame size: 13.3 Kbps, 30ms frame size
- **Speex** - 2.15 to 44.2 Kbps
- **LPC10** - 2.5 Kbps
- **H.261 Video** - Used over ISDN lines with resolution of 352x288
- **H.263 Video** - Low-bit rate encoding solution for video conferencing
- **H.263+ Video** - Extension of H.263 that provides additional features that improve compression over packet switched networks.

21.1.13 Monitoring

Monitoring sets alarms and notifications at which the system will monitor itself for normal operation and where by appropriate notifications are sent if alarms are triggered.

NOTE: Reloading the system will not interrupt any services while restarting the system does stop *and* starts all system services.



Monitor (mins):

Time interval at which system should check if Asterisk is down. If down, system will try to start it and will send a notification email about the stop/start action

Example: 15
Field Type: [0-9]

Reload Type:

Select whether to reload the system at some specific time of a day or in regular time intervals (hourly)

Example: Setting this option to 'Time of the day' and 'Reload (hours)' = '2' will reload the system every day at 02:00 hours. Setting this option to 'Regular Interval' and 'Reload (hours)' = '2' will reload the system every two hours
Field Type: Select box

Reload (hours):

This field is active only when 'Reload Type' option is selected

Example: Setting 'Reload Type' = 'Time of the day' and this option to '2' will reload the system every day at 02:00 hours. Setting 'Reload Type' = 'Regular Interval' and this option to '2' will reload the system every two hours
Field Type: Select box

Restart Type:

Select whether to restart the system at some specific time of a day or in regular time intervals (hourly)

Example: Setting this option to 'Time of the day' and 'Restart (hours)' = '2' will restart the system every day at 02:00 hours. Setting this option to 'Regular Interval' and 'Restart (hours)' = '2' will restart the system every two hours
Field Type: Select box

Restart (hours):

This field is active only when 'Restart Type' option is selected

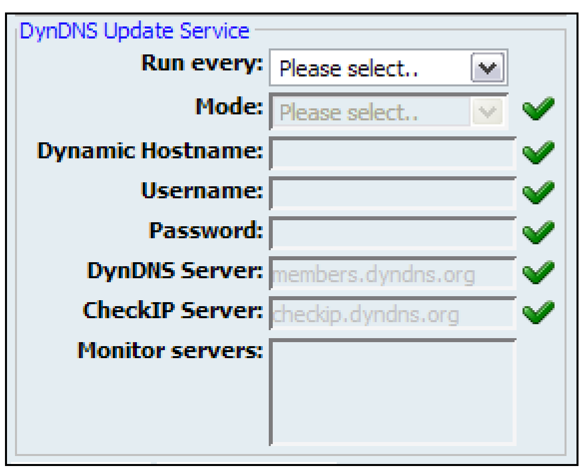
Example: Setting 'Restart Type' = 'Time of the day' and this option to '2' will restart the system every day at 02:00 hours. Setting 'Restart Type' = 'Regular Interval' and this option to '2' will restart the system every two hours
Field Type: Select box

Notification e-mail:

Email address on which reload/restart notification is sent

Example: email@domain.com
Field Type: [a-z][0-9]

21.1.14 DynDNS Update Service



Run every:

Select time interval at which the DynDNS update request will be sent

Example: Select 'minute' to send request to DynDNS server every minute
Field Type: Select box

Mode:

Set the way DynDNS service will work on the system

Example: 'Update & Monitor' will monitor and update/reload IP address(es) of monitored servers once they change. 'Monitor only' will monitor for IP address change but will not update/reload
Field Type: Select box

Dynamic Hostname:

Hostname to which requests are periodically sent to

Example: domain.dyndns.org
Field Type: [a-z][0-9]

Username:

Username for DynDNS authentication

Example: Type username used for DynDNS authentication here
Field Type: [a-z][0-9]

Password:

Password for DynDNS authentication

Example: Type password used for DynDNS authentication here
Field Type: [a-z][0-9]

DynDNS Server:

DynDNS server name

Example: members.dyndns.org, (default value)
Field Type: [a-z][0-9]

CheckIP Server:

DynDNS server for checking the IP addresses

Example: checkip.dyndns.org, (default value)
Field Type: [a-z][0-9]

Monitor Servers:

Dyndns domains of monitored servers separated by a blank space

Example: serverone.dyndnd.org
Field Type: [a-z][0-9]

21.2 Protocols

Description:
Protocol is a set of rules that allows UAD, systems, networks etc. to communicate using a set standard.
Supported protocols are:

- SIP 377
- IAX 391
- MGCP 400

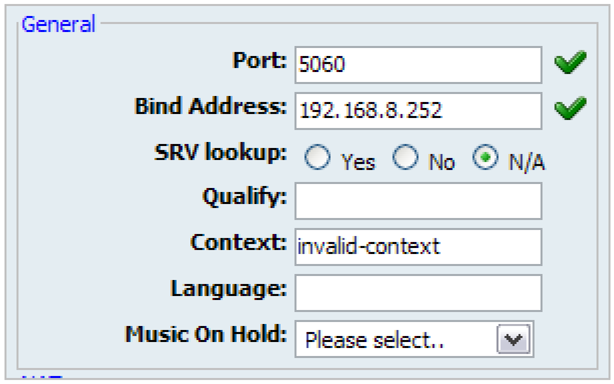
21.2.1 SIP

SIP (**S**ession **I**nitiated **P**rotocol, or **S**ession **I**nitiation **P**rotocol), is a signaling protocol for Internet conferencing, telephony, presence, events notification and instant messaging. The protocol initiates call setup, routing, authentication and other feature messages to end points within an IP domain.

In this chapter we will cover:

- General 378
- NAT 379
- Security 380
- RTP 382
- DTMF 382
- Misc 383
- Authentication 385
- Registration 386
- MWI 387
- Subscriptions 388
- Domains 388
- Codecs 390
- Additional config 391

21.2.1.1 General



Port:

SIP bind port

Example: 5060, (default)
Field Type: [0-9]

Bind Address:

SIP bind IP address

Example: 0.0.0.0, (default)
Field Type: [0-9]

SVR lookup:

Enable DNS SRV lookups on outbound calls

Example: Disabling this option will disable SIP calls based on domain names between SIP users on the Internet
Field Type: Option buttons

Qualify:

Timing interval in milliseconds at which a 'ping' is sent to a host in order to find out its status

Example: Set this field to 2000 for example. If more time then provided here is needed to reach the host,

host is considered offline
Field Type: [0-9]

Context:

Default context for incoming calls

Example: For security reasons it is recommended to keep this field set at 'invalid-context'
Field Type: [a-z][0-9]

Language:

Default language settings for all users/peers

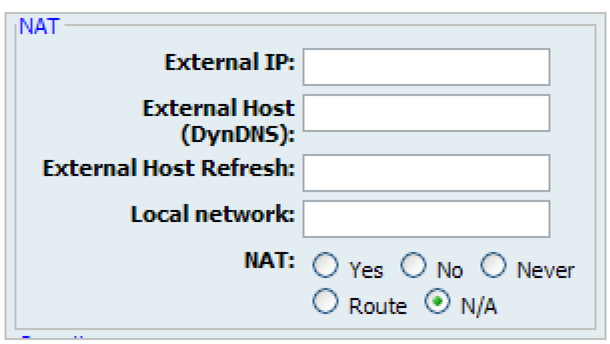
Example: Set this option to 'en' (English) for example
Field Type: [a-z]

Music on Hold:

Set the default MOH (Music on Hold) class for all SIP calls

Example: Set 'default' for example to play 'default' MOH class to all SIP calls when placed on hold for example
Field Type: Select box

21.2.1.2 NAT



External IP:

External IP/Public/Internet address system uses

Example: If your system is behind NAT set this option to Public/Internet IP address system uses when registering with other proxies over Internet
Field Type: [0-9]

External Host (DynDNS):

DynDNS address system uses

Example: If your system is behind NAT, along with the External IP address you may use the DynDNS service as well. Set this field to DynDNS host
Field Type: [0-9]

External Host Refresh:

How often to refresh External DynDNS host (if used)

Example: Time in seconds (e.g. 10)
Field Type: [0-9]

Local network:

If system is used in local network, set the local network address here

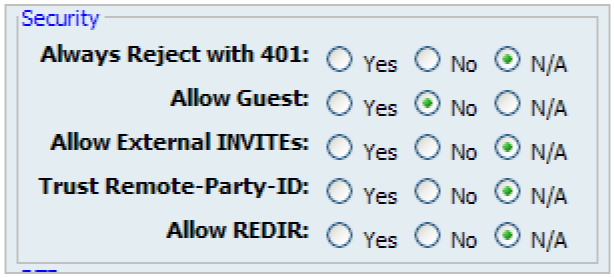
Example: 192.168.0.0/255.255.0.0
Field Type: [0-9]

NAT:

Global SIP NAT setting which affects all users/peers

Example: Set this option to 'Yes' if system is behind NAT
Field Type: Option buttons

21.2.1.3 Security



Always Reject with 401:

Example:
Field Type: Option buttons

Allow guest:

Example:
Field Type: Option buttons

Allow External INVITEs:

Example:
Field Type: Option buttons

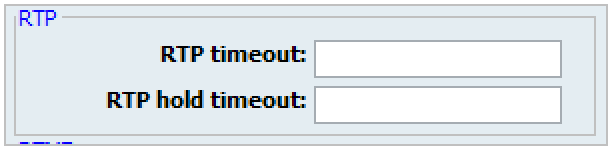
Trust Remote-Party-ID:

Example:
Field Type: Option buttons

Allow REDIR:

Example:
Field Type: Option buttons

21.2.1.4 RTP



RTP timeout:

Max RTP timeout

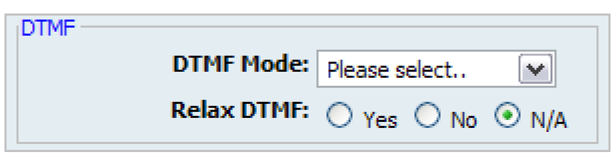
Example: All calls (if not on hold) will be terminated if there is no RTP activity for number of seconds set here (60 for example)
Field Type: [0-9]

RTP hold timeout:

Max RTP hold timeout. NOTE: This field must be higher number then set under 'RTP timeout'

Example: All calls on hold will be terminated if there is no RTP activity for number of seconds set here (300 for example)
Field Type: [0-9]

21.2.1.5 DTMF



DTMF Mode:

Set the default DTMF mode

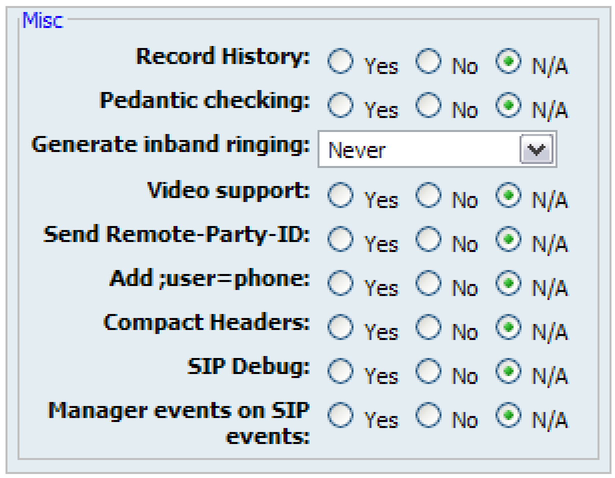
Example: rfc2833
Field Type: Select box

Relax DTMF:

Relax DTMF handling

Example: Set this field to 'Yes' if having problems with DTMF modes
Field Type: Option buttons

21.2.1.6 Misc



Record History:

Should SIP history be recorded

Example: Select 'Yes' to record SIP history. Example history information:

* SIP Call
1. TxReqRel INVITE / 102 INVITE
2. Rx SIP/2.0 / 102 INVITE /100 Trying
3. CancelDestroy
4. Rx SIP/2.0 / 102 INVITE /180 Ringing
5. CancelDestroy
6. Rx SIP/2.0 / 102 INVITE /200 OK
7. CancelDestroy
8. Unhold SIP/2.0
9. TxReq ACK / 102 ACK
10. TxReqRel INVITE / 103 INVITE
11. Rx SIP/2.0 / 103 INVITE /200 OK
12. CancelDestroy
13. Unhold SIP/2.0
14. TxReq ACK / 103 ACK

Field Type: Option buttons

Pedantic checking:

Enable slow, pedantic checking for Pingtel and multi-line formatted headers for strict SIP compatibility

Example: It is recommended to set this field to 'No'
Field Type:Option buttons

Generate inband ringing:

Set whether the system generates in-bank ringing

Example: You're recommended to set this option to 'Never'
Field Type: Select box

Video support:

Set whether the system generates in-bank ringing

Example: You're recommended to set this option to 'No'
Field Type: Option buttons

Send Remote-Party-ID:

Should 'Remote-Party-ID' be added to uri

Example: You're recommended to set this option to 'No' unless required otherwise
Field Type: Option buttons

Add ;user=phone:

Should ';user=phone' be added ot uri

Example: You're recommended to set this option to 'No' unless required otherwise
Field Type: Option buttons

Compact Headers:

Should compact SIP headers be sent

Example: You're recommended to set this option to 'No' unless required otherwise
Field Type: Option buttons

SIP Debug:

Should SIP Debug be turned on all the time

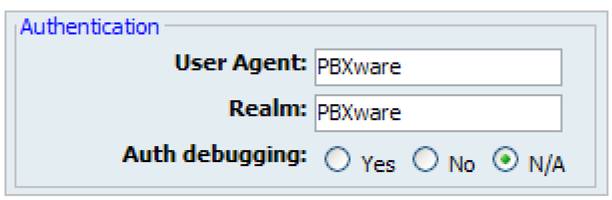
Example: You're recommended to set this option to 'No' unless required otherwise
Field Type: Option buttons

Manager events on SIP events:

Should manager events be generated if SIP UAD/Phone performs some event (Hold for example)

Example: You're recommended to set this option to 'No' unless required otherwise
Field Type: Option buttons

21.2.1.7 Authentication



User Agent:

Set the 'User Agent' string

Example: 'Custom string' for example
Field Type: [a-z][0-9]

Realm:

Realm for digest authentication

Example: 'Custom string' for example

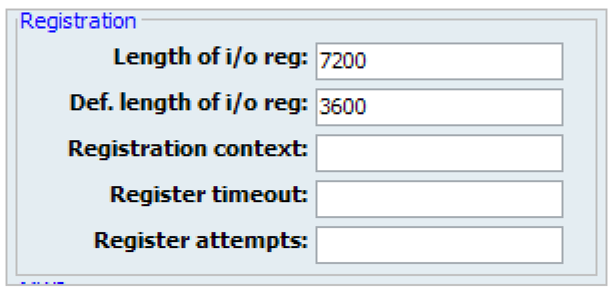
Field Type: [a-z][0-9]

Auth debugging:

Should authentication be debugged

Example: Setting this option to 'Yes' will increase the amount of debugging traffic
Field Type: Option buttons

21.2.1.8 Registration



Length of i/o reg:

Example:
Field Type: [0-9]

Def. Length of i/o reg:

Example:
Field Type: [0-9]

Registration context:

Should system dynamically create and destroy noop priority 1 extension for peer who (un)registers with us

Example: sipregistrations
Field Type: [a-z][0-9]

Registration timeout:

Number of seconds after which registration times out

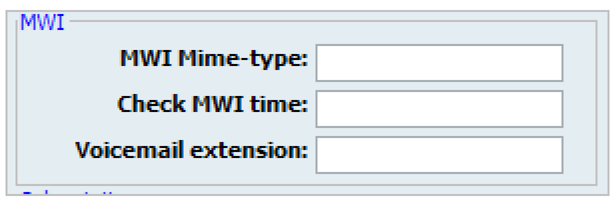
Example: Default value 20
Field Type: [0-9]

Register attempts:

Number of registration attemps

Example: One 'Register timeout' equals one 'Registration attempts'. Default value 10
Field Type: [0-9]

21.2.1.9 MWI



MWI Mime-type:

Allow overriding of mime type

Example: Default value 'text/plain'
Field Type: [a-z]

Check MWI time:

Default time between mailbox checks for peers

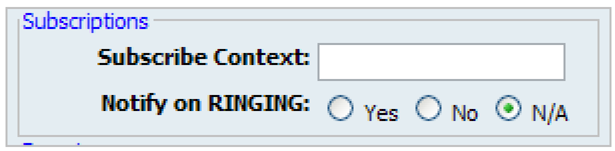
Example: Default value 10
Field Type: [0-9]

Voicemail extension:

Dialplan extension to reach mailbox. This option sets the 'Message-Account' in the MWI notify message

Example: Default value 'asterisk'
Field Type: [a-z][0-9]

21.2.1.10 Subscriptions



Subscribe Context:

Set a specific context for SUBSCRIBE requests (Useful to limit subscriptions to local extensions)

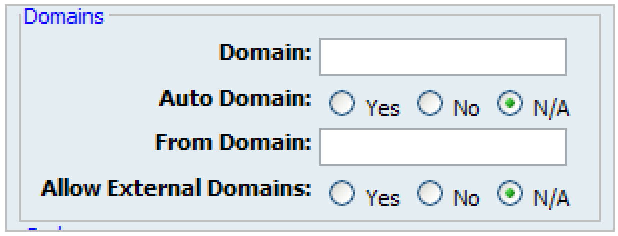
Example: -
Field Type: [a-z][0-9]

Notify on RINGING:

Notify subscriptions on RINGING state

Example: -
Field Type: Option buttons

21.2.1.11 Domains



Domain:

Set default domain for this host

Example: If configured, Asterisk will only allow INVITE and REFER to non-local domains. Use 'sip show domains' to list local domains
Field Type: [a-z][0-9]

Auto Domain:

Turn this on to have Asterisk add local host name and local IP to domain list.

Example: If system host name is set to 'my_system', with this feature set to 'On', 'my_system' will be automatically added to domain list
Field Type: Option buttons

From Domain:

Change the 'From: ' headers

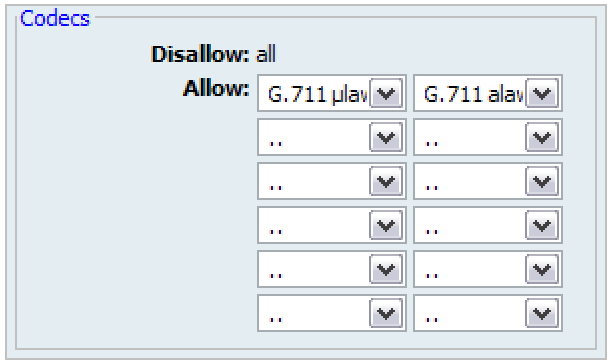
Example: Keep this field empty unless requested otherwise
Field Type: [a-z][0-9]

Allow External Domains:

Should domains not serviced by this server be (dis)allowed

Example:
Field Type: Option buttons

21.2.1.12 Codecs



Disallow:

Set the codecs extension is now allowed to use

Example: This field is very unique. In order to work properly, this setting is automatically set to 'Disallow All' and it cannot be modified
Field Type: Read only

Allow:

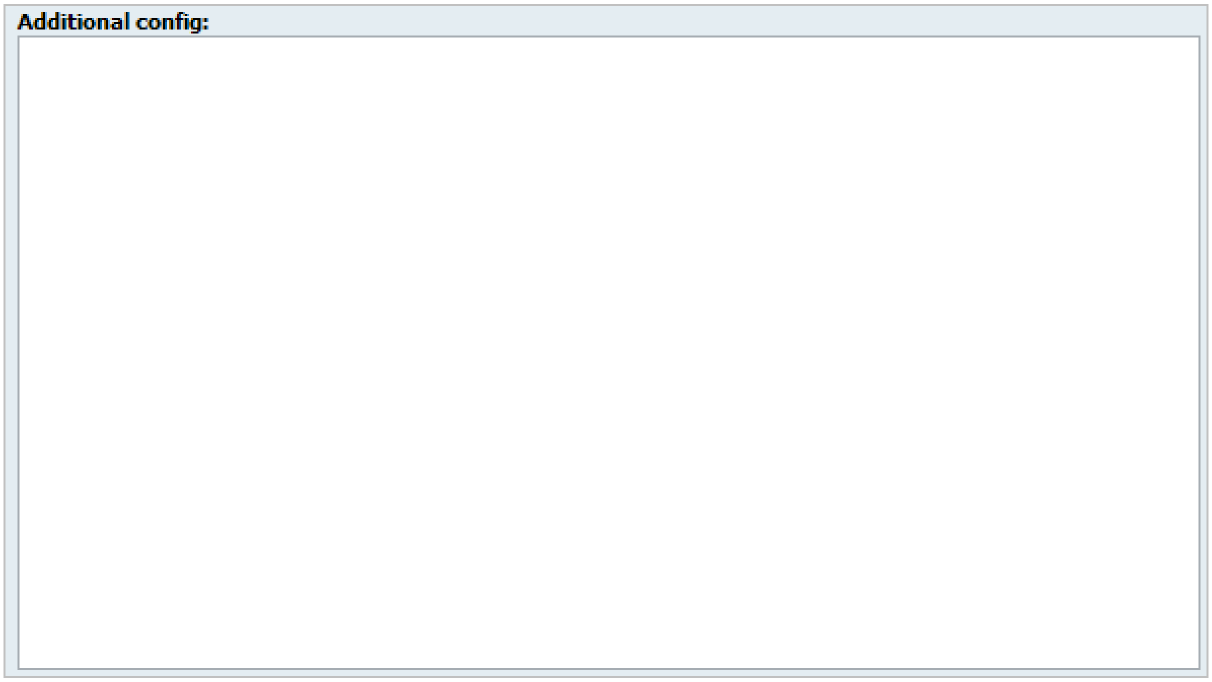
Set the codecs extension is allowed to use

Example: Only the codecs set under 'Settings: Server' will be available to choose from
Field Type: Check box

- **ITU G.711 ulaw** - 64 Kbps, sample-based, used in US
- **ITU G.711 alaw** - 64 Kbps, sample-based, used in Europe
- **ITU G.723.1** - (5.3/6.3 Kbps, 30ms frame size
- **ITU G.726** - 16/24/32/40 Kbps
- **ITU G.729** - 8 Kbps, 10ms frame size
- **GSM** - 13 Kbps (full rate), 20ms frame size
- **iLBC** - 15Kbps,20ms frame size: 13.3 Kbps, 30ms frame size
- **Speex** - 2.15 to 44.2 Kbps
- **LPC10** - 2.5 Kbps
- **H.261 Video** - Used over ISDN lines with resolution of 352x288
- **H.263 Video** - Low-bit rate encoding solution for video conferencing
- **H.263+ Video** - Extension of H.263 that provides additional features that improve compression over packet switched networks.

21.2.1.13 Additional config

This option is used for providing additional config parameters for SIP configuration files.
Values provided here will be written into these configuration files.

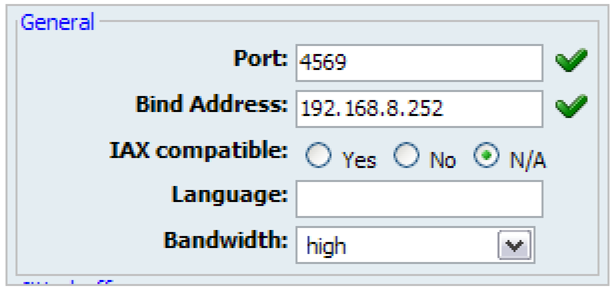


21.2.2 IAX

IAX (Inter asterisk exchange) is a simple, low overhead and low bandwidth VoIP protocol designed to allow multiple PBXwares to communicate with one another without the overhead of more complex protocols. Payload is sent with

In this chapter we will cover:

21.2.2.1 General



Port:

SIP bind port

Example: 4569, (default)
Field Type: [0-9]

Bind Address:

SIP bind IP address

Example: 0.0.0.0, (default)
Field Type: [0-9]

IAX compatible:

Should layered switches or some other scenario be used

Example: Set to yes if you plan to use layered switches or some other scenario which may cause some delay when doing a lookup in the dialplan
Field Type: Select box

Language:

Default language settings for all users/peers

Example: Set this option to 'en' (English) for example
Field Type: [a-z]

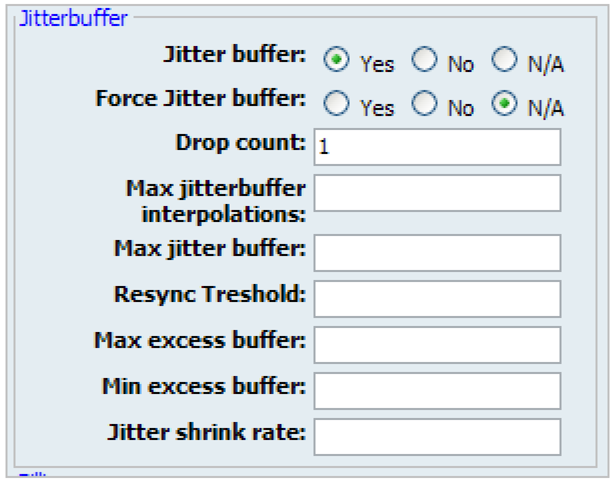
Bandwidth:

Set the bandwidth to control which codecs are used in general

Example: Select between low, mid or high

Field Type: Select box

21.2.2.2 Jitterbuffer



Jitter Buffer:

Turn off jitter buffer for this peer

Example: Yes, No, N/A
Field Type: Option buttons

Force Jitter Buffer:

Should we force jitter buffer (default value 10)

Example: Jitter buffer is usually handled by the UADs/Phones. But in case if these do this poorly jitter buffer can be enforced on PBXware side
Field Type: [0-9]

Drop count:

Set number of frames that can be dropped over the last 2 seconds. Set small number cause 3 = 1.5% of frames dropped

Example: 1
Field Type: [0-9]

Max. jitterbuffer interpolations:

The maximum number of interpolation frames the jitterbuffer should return in a row

Example: 1000
Field Type: [0-9]

Max. Jitter buffer:

A maximum size for the jitter buffer. Setting a reasonable maximum here will prevent the call delay from rising to silly values in extreme situations; you'll hear SOMETHING, even though it will be jittery.

Example: 1000
Field Type: [0-9]

Resync Treshold:

Resync the threshold for noticing a change in delay measured

Example: 1000
Field Type: [0-9]

Max Excess Buffer:

Maximum amount of headroom in the jitter buffer

Example: 80
Field Type: [0-9]

Min Excess Buffer:

Minimum amount of headroom in the jitter buffer

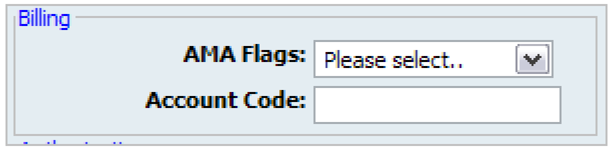
Example: 10
Field Type: [0-9]

Jitter Shrink Rate:

How many milli seconds shall be taken off per 20ms frame received

Example: 1
Field Type: [0-9]

21.2.2.3 Billing



AMA Flags:

These flags are used in the generation of call detail records (e.g 'default')

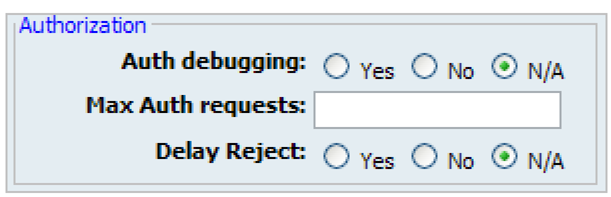
Example: Select between 'default', 'omit', 'billing' or 'documentation'
Field Type: Select box

Account code:

Default account for CDRs (Call Detail Records)

Example: lars101
Field Type: [a-z][0-9]

21.2.2.4 Authorization



Auth debugging:

Should authentication be debugged

Example: Setting this option to 'Yes' will increase the amount of debugging traffic
Field Type: Option buttons

Max Auth requests:

Maximum number of outstanding authentication requests waiting for replies. Any further authentication attempts will be blocked

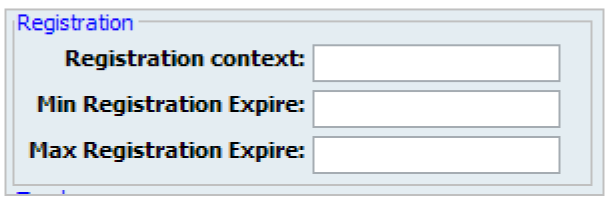
Example: 10
Field Type: [0-9]

Delay Reject:

Set this option to 'Yes' for increased security against brute force password attacks

Example: Yes
Field Type: [0-9]

21.2.2.5 Registration



Registration context:

If specified PBXware will dynamically create and destroy a NoOp priority 1 extension for a given peer who registers or unregisters with us

Example: iaxregistration
Field Type: [a-z][0-9]

Min Registration Expire:

Minimum amounts of time that IAX peers can request as a registration expiration interval (in seconds).

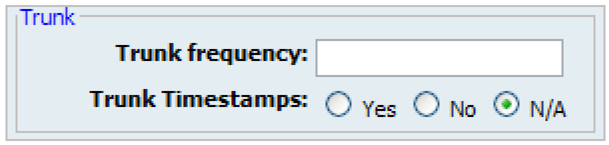
Example: 60
Field Type: [0-9]

Max Registration Expire:

Maximum amounts of time that IAX peers can request as a registration expiration interval (in seconds).

Example: 60
Field Type: [0-9]

21.2.2.6 Trunk



Trunk frequency:

How frequently to send trunk msgs (in ms)

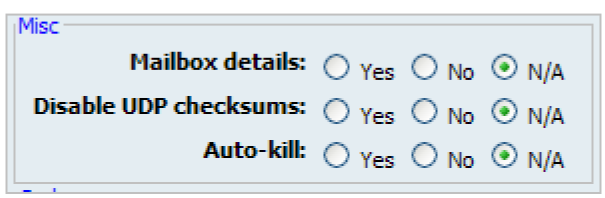
Example: 20
Field Type: [0-9]

Trunk Timestamps:

Should we send timestamps for the individual sub-frames within trunk frames

Example: Yes
Field Type: Option buttons

21.2.2.7 Misc



Mailbox details:

Should the user receive the actual new/old message counts and not just a yes/no messages

Example: Yes
Field Type: Option buttons

Disable UDP checksums:

Should checkums will be calculated

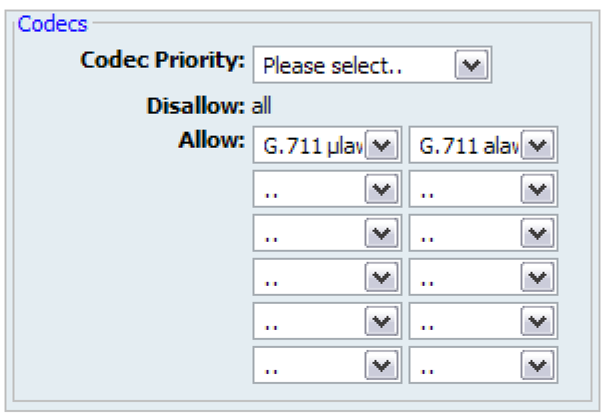
Example: Yes
Field Type: Option buttons

Auto-kill:

If no response is received within 2000ms, and this option set to yes, cancel the whole thing

Example: Yes
Field Type: Option buttons

21.2.2.8 Codecs



Codec Priority:

This option controls the codec negotiation of an inbound IAX calls.

Example:
caller - Consider the callers preferred order ahead of the host's.
host - Consider the host's preferred order ahead of the caller's.
disabled - Disable the consideration of codec preference altogether (this is the original behaviour before preferences were added)
reqonly - Same as disabled, only do not consider capabilities if the requested format is not available the call will only be accepted if the requested form
Field Type: Read only

Disallow:

Set the codecs extension is now allowed to use

Example: This field is very unique. In order to work properly, this setting is automatically set to 'Disallow All' and it cannot be modified
Field Type: Read only

Allow:

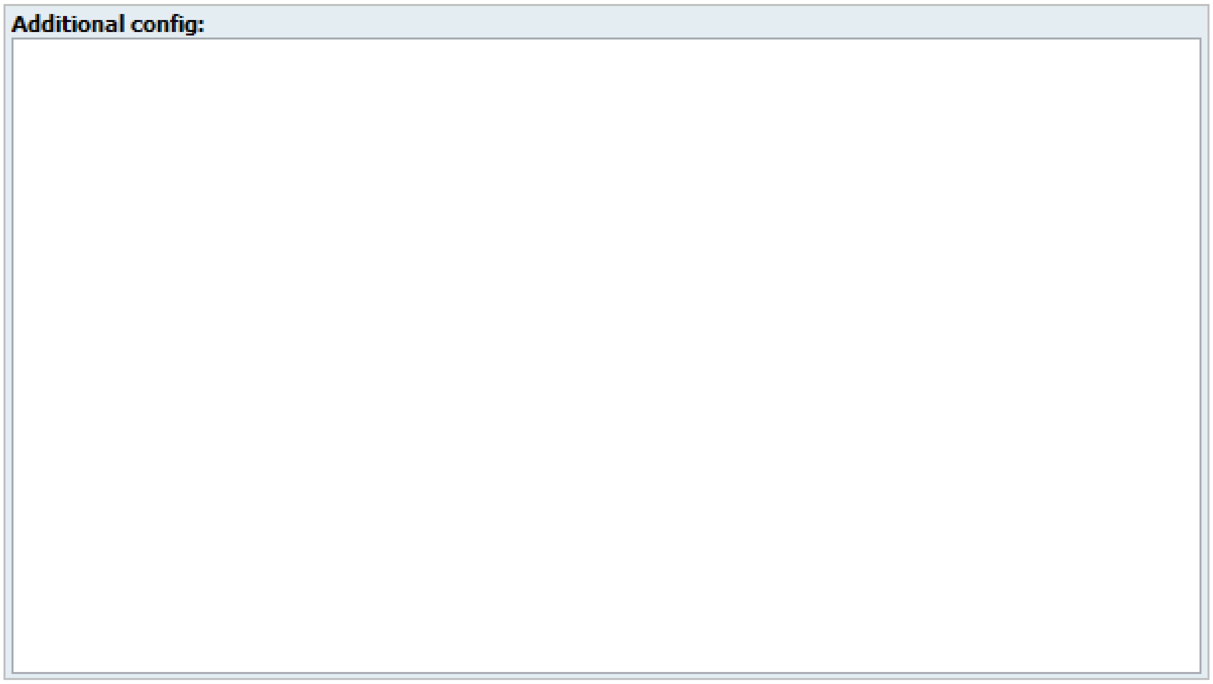
Set the codecs extension is allowed to use

Example: Only the codecs set under 'Settings: Server' will be available to choose from
Field Type: Check box

- **ITU G.711 ulaw** **-** 64 Kbps, sample-based, used in US
- **ITU G.711 alaw** - 64 Kbps, sample-based, used in Europe
- **ITU G.723.1** - (5.3/6.3 Kbps, 30ms frame size
- **ITU G.726** - 16/24/32/40 Kbps
- **ITU G.729** - 8 Kbps, 10ms frame size
- **GSM** - 13 Kbps (full rate), 20ms frame size
- **iLBC** - 15Kbps,20ms frame size: 13.3 Kbps, 30ms frame size
- **Speex** - 2.15 to 44.2 Kbps
- **LPC10** - 2.5 Kbps
- **H.261 Video** - Used over ISDN lines with resolution of 352x288
- **H.263 Video** - Low-bit rate encoding solution for video conferencing
- **H.263+ Video** - Extension of H.263 that provides additional features that improve compression over packet switched networks.

21.2.2.9 Additional config

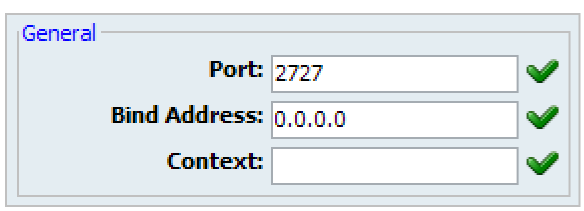
This option is used for providing additional config parameters for IAX configuration files.
Values provided here will be written into these configuration files.



21.2.3 MGCP

MGCP is a protocol for controlling Telephony Gateways from external call control elements named Media Gateway Controllers or Call Agents. MGCP is central to the VoIP solution and may be integrated into products such as Central Office Switches, Gateways (Trunking, Residential, Access), Network Access Servers, Cable Modems, PBXwares etc., to develop a convergent voice and data solution.

21.2.3.1 General



Port:

SIP bind port

Example: 2727, (default)
Field Type: [0-9]

Bind Address:

SIP bind IP address

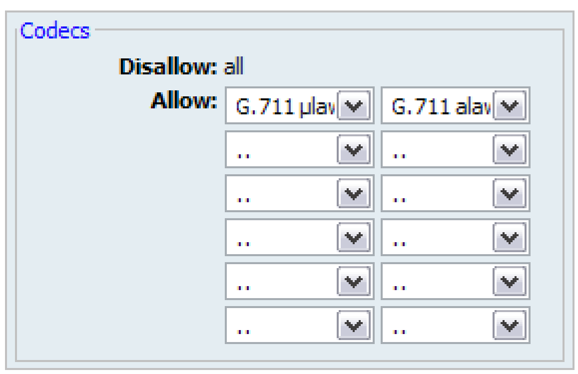
Example: 0.0.0.0, (default)
Field Type: [0-9]

Context:

Default context for incoming calls

Example: For security reasons it is recommended to keep this field set at 'invalid-context'
Field Type: [a-z][0-9]

21.2.3.2 Codecs



Disallow:

Set the codecs extension is now allowed to use

Example: This field is very unique. In order to work properly, this setting is automatically set to 'Disallow All' and it cannot be modified
Field Type: Read only

Allow:

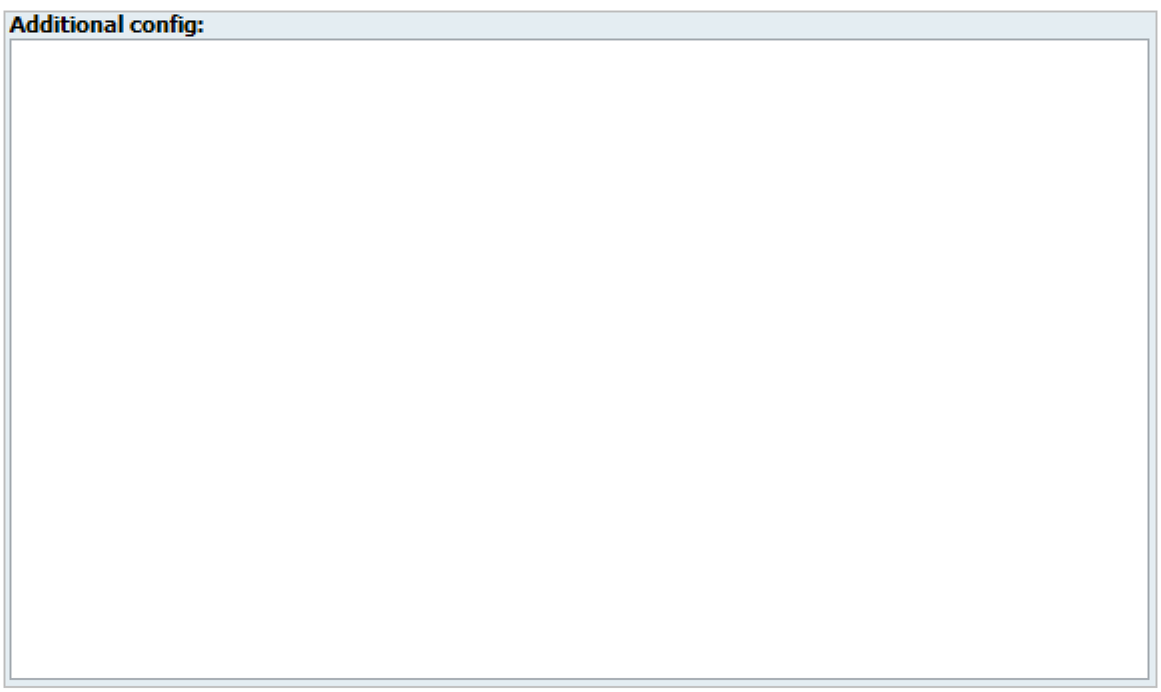
Set the codecs extension is allowed to use

Example: Only the codecs set under 'Settings: Server' will be available to choose from
Field Type: Check box

- **ITU G.711 ulaw** - 64 Kbps, sample-based, used in US
- **ITU G.711 alaw** - 64 Kbps, sample-based, used in Europe
- **ITU G.723.1** - (5.3/6.3 Kbps, 30ms frame size
- **ITU G.726** - 16/24/32/40 Kbps
- **ITU G.729** - 8 Kbps, 10ms frame size
- **GSM** - 13 Kbps (full rate), 20ms frame size
- **iLBC** - 15Kbps,20ms frame size: 13.3 Kbps, 30ms frame size
- **Speex** - 2.15 to 44.2 Kbps
- **LPC10** - 2.5 Kbps
- **H.261 Video** - Used over ISDN lines with resolution of 352x288
- **H.263 Video** - Low-bit rate encoding solution for video conferencing
- **H.263+ Video** - Extension of H.263 that provides additional features that improve compression over packet switched networks.

21.2.3.3 Additional config

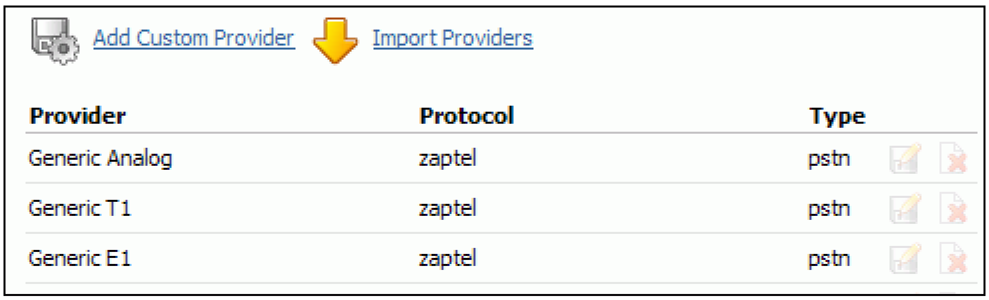
This option is used for providing additional config parameters for MGCP configuration files.
Values provided here will be written into these configuration files.



21.3 Providers

PBXware comes with a range of pre-configured VoIP and PSTN service providers in order to allow an easy way of adding
trunks into the system. This screen allows for addition of custom providers by clicking on 'Add Custom Provider'.

In addition, 'Import Providers' allows for update of currently pre-configured service providers.



Provider	Protocol	Type
Generic Analog	zaptel	pstn
Generic T1	zaptel	pstn
Generic E1	zaptel	pstn

Provider:
Provider name
Example: Generic Analog
Field Type: Display

Protocol:
Protocol provider uses
Example: zaptel
Field Type: Display

Type:
Service type
Example: pstn/voip
Field Type: Display


Edit Provider configuration
Example: Click to edit Provider configuration
Field Type: Button


Delete Provider configuration
Example: Click to delete Provider configuration from the system
Field Type: Button

21.3.1 PSTN

Type:

Service type

Example: pstn/voip
Field Type: Select box

Protocol:

Protocol provider uses

Example: zaptel/capi
Field Type: Select box

Provider:

Provider name

Example: BT
Field Type: [a-z][0-9]

Country:

Destination of the trunk connection

Example: USA
Field Type: Select box

E164 Accepted:

Does the Provider support dialing destinations in E164 format

Example: Enabling this option will reformat any dialled number into following form COUNTRY_CODE +AREA_CODE+DIALED_NUMBER. For example, if you dial 55510205, system will dial 121255510205
Field Type: Option buttons

National Dialing Code:

National dialing code at the Provider destination

Example: For USA **1**, United Kingdom, Germany **0**
Field Type: [0-9]

Leave National Code:

In some countries, national code is stripped automatically. If set to 'Yes', national code will not be stripped from the dialed number. NOTE: Before settings this option to 'Yes', go to 'Settings: Servers' and enable this options as well.

Example: John dials 121255510205. With this option enabled
Field Type: [0-9]

International Dialing Code:

International dialing code at the Provider destination

Example: For USA **011**, United Kingdom, Germany **00**
Field Type: [0-9]

Local Area Code:

Add local area code to dialed number, if required by service provider. (By default, local area code is stripped when dialing)

Example: User dials 55510205, local area code is 212. If call goes through this trunk PBXware will dial 21210205
Field Type: [0-9]

Write dialing code:

Should National and International prefix be written into configuration files

Example: Enable this option if required by provider
Field Type: [0-9]

Zaptel Devices:

Select zaptel device system is to use

Example:

- X100P
- TDM400P
- TDM10B
- TDM20B
- TDM30B
- TDM40B
- TDM11B
- TDM12B
- TDM13B
- TDM21B
- TDM22B
- TDM23B
- T100P
- E100P
- TE110P

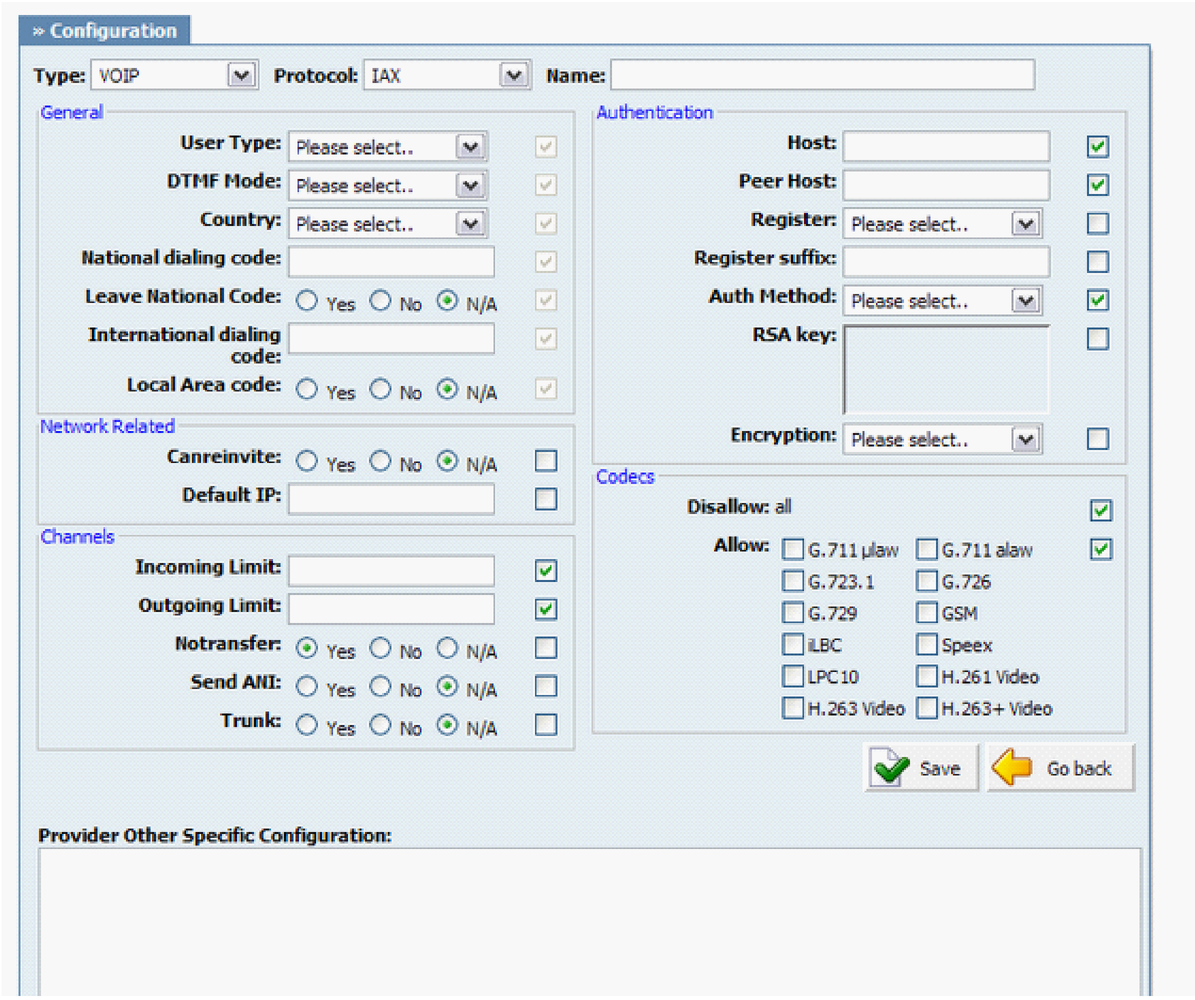
- TE410P
- TE405P
- hfcISDN
- quadBRI
- A101u
- A102u
- A104u

Field Type: Option buttons

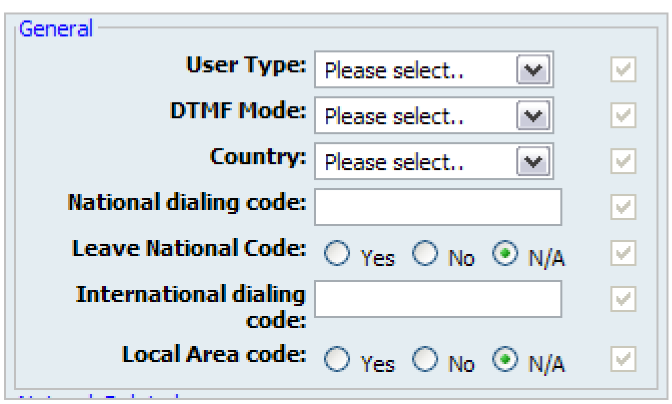
NOTE: Please configure Zaptel devices after you save the provider by clicking 'Configure' button next to a selected device.

21.3.2 VoIP

In this chapter we will cover:



21.3.2.1 General



Type:

Service type

Example: pstn/voip
Field Type: Select box

Protocol:

Protocol provider uses

Example: zaptel/capi
Field Type: Select box

Provider:

Provider name

Example: BT
Field Type: [a-z][0-9]

User Type:

User's relationship to the system

Example:
- **user** - Provider accepts incoming calls only
- **peer** - Provider makes outgoing calls only
- **friend** - Provider does both incoming and outgoing calls

Field Type: Select box

DTMF Mode (Dual Tone Multi-Frequency):

DTMF mode used by provider. A specific frequency (consisting of two separate tones) to each key so that it can easily be identified by a microprocessor

Example:
- **inband** - inband audio(requires 64 kbit codec - alaw, ulaw)
- **rfc2833** - default
- **info** - SIP INFO messages

Field Type: Select box

National Dialing Code:

National dialing code used at the Provider destination

Example: For USA **1**, United Kingdom, Germany **0**
Field Type: [0-9]

Leave National Code:

In some countries, national code is stripped automatically. If set to 'Yes', national code will not be stripped from the dialed number. NOTE: Before settings this option to 'Yes', go to 'Settings: Servers' and enable this options as well.

Example: John dials 121255510205. With this option enabled
Field Type: [0-9]

International Dialing Code:

International dialing code used at the provider destination

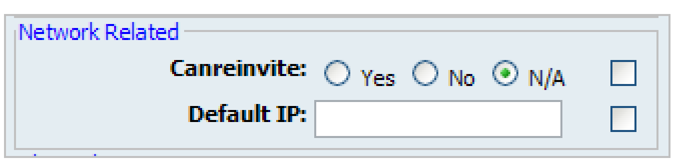
Example: For USA **011**, United Kingdom, Germany **00**
Field Type: [0-9]

Local Area Code:

Add local area code to dialed number, if required by service provider. (By default, local area code is stripped when dialing)

Example: User dials 55510205, local area code is 212. If call goes through this provider PBXware will dial 21210205
Field Type: [0-9]

21.3.2.2 Network Related



Canreinvite:

Should you allow RTP voice traffic to bypass Asterisk

Example: Some devices do not support this especially if one of them is behind a NAT
Field Type: Options buttons

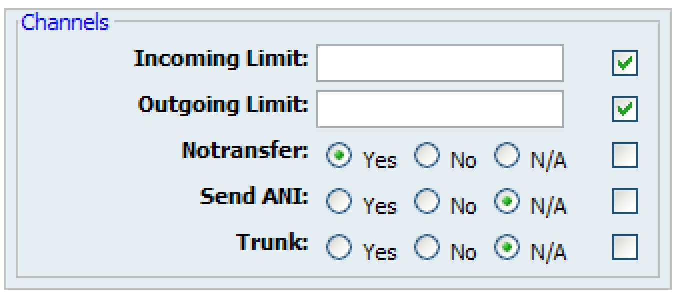
NOTE: All enhanced services for the extension have to be disabled

Default IP:

IP address to be used until registration

Example: 192.168.1.1
Field Type: IP Address

21.3.2.3 Channels



Incoming Limit:

Number of simultaneous incoming calls Provider can handle

Example: 4 equals to four simultaneous incoming calls. Any additional calls will get the busy sound
Field Type: [0-9]

Outgoing Limit:

Number of simultaneous outgoing calls Provider can handle

Example: 4 equals to four simultaneous outgoing calls. Any additional calls attempting to use this Provider will be rejected or will be redirected to other Providers depending on what is set in the system/extensions
Field Type: [0-9]

Notransfer:

Disable native IAX transfer

Example:
Field Type: Option buttons

Send ANI:

Should ANI ("super" Caller ID) be sent over this Provider

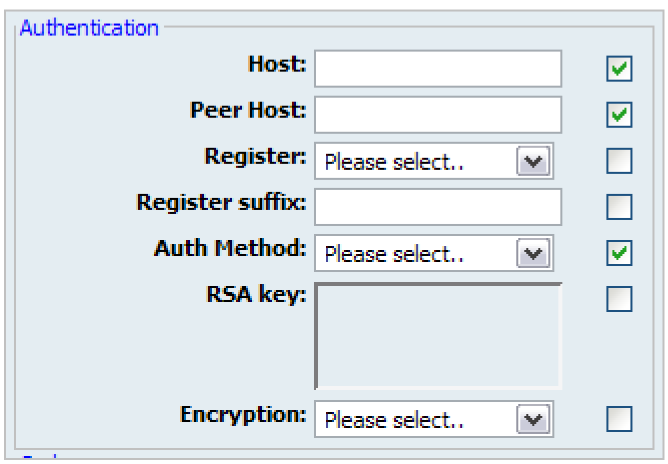
Example: Set 'Yes' to enable
Field Type: Option buttons

Trunk:

Use IAX2 trunking with this host

Example: Set 'Yes' to enable
Field Type: Option buttons

21.3.2.4 Authentication



Host:

Provider IP address

Example: Enter host IP, 192.168.1.1 for example or set 'dynamic' if host is behind dynamic IP address
Field Type: [0-9][a-z]

Peer Host:

IP of a peer host system sends the calls to

Example: 192.168.1.1
Field Type: IP Address

Register:

Method for registering to remote server

Example: Providers may require different way of registration to their server. You may choose between 'registration not required', 'register with phone number' and 'register with username'
Field Type: Select box

Register suffix:

Service provider may request different registration methods for their services. Select the proper method, as required by the provider

Example: 1234567
Field Type: [0-9]

Auth Method:

Authentication method required by provider

Example: md5
Field Type: [a-z] [0-9]

RSA key:

RSA authentication key

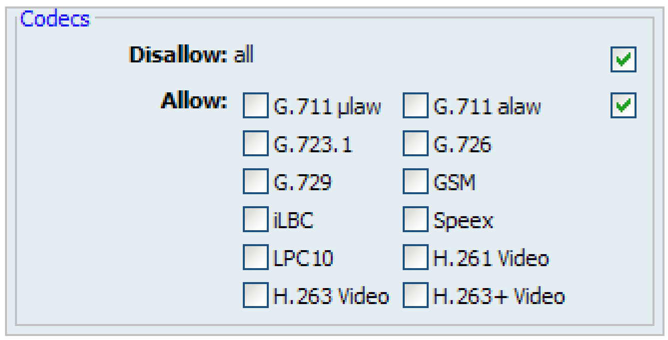
Example: If Auth Method is set to RSA, then provide the RSA key here
Field Type: [a-z][0-9]

Encryption:

Should encryption be used when authenticating with the peer

Example:
Field Type: [a-z][0-9]

21.3.2.5 Codecs



Disallow:
This field is very unique. In order to work properly this setting is automatically set to **'Disallow All'** and cannot be modified.
Example: all
Field Type: Display

Allow:
Codecs that are allowed in **'Settings: Server'** will be enabled for selection.
Example: all

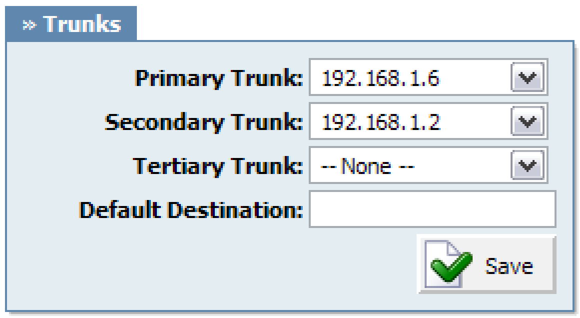
- **ITU G.711 ulaw** **-** 64 Kbps, sample-based, used in US
- **ITU G.711 alaw** - 64 Kbps, sample-based, used in Europe
- **ITU G.723.1** - (5.3/6.3 Kbps, 30ms frame size
- **ITU G.726** - 16/24/32/40 Kbps
- **ITU G.729** - 8 Kbps, 10ms frame size
- **GSM** - 13 Kbps (full rate), 20ms frame size
- **iLBC** - 15Kbps,20ms frame size: 13.3 Kbps, 30ms frame size
- **Speex** - 2.15 to 44.2 Kbps
- **LPC10** - 2.5 Kbps
- **H.261 Video** - Used over ISDN lines with resolution of 352x288
- **H.263 Video** - Low-bit rate encoding solution for video conferencing
- **H.263+ Video** - Extension of H.263 that provides additional features that improve compression over packet switched networks.

Field Type: Check boxes

21.4 Default Trunks

System uses its trunks to place calls to various destinations. In order to allow an organization to control its voice communications budget and to provide for termination backup the default trunks allows setting primary, secondary and tertiary trunks.

System will use primary trunk as its first choice for every destination called. If primary trunk for same reason fails to terminate the call, the secondary trunk wild be used by the system. If secondary trunk for same reason fails to terminate the call, the tertiary trunk wild be used by the system.



Primary/Secondary/Tertiary Trunk:

Select default trunks on system level.

NOTE: If dialed number is busy, the system will recognize it and won't skip to other trunks in order to dial it.

Example: Service provider name
Field Type: Select box

Default Destination:

Default destination where trunks without DID will be transferred to

Example: If there is no DID for an incoming Trunk/Provider, all calls will be transferred to this system number (e.g. 2000)
Field Type: [0-9]

Precedence

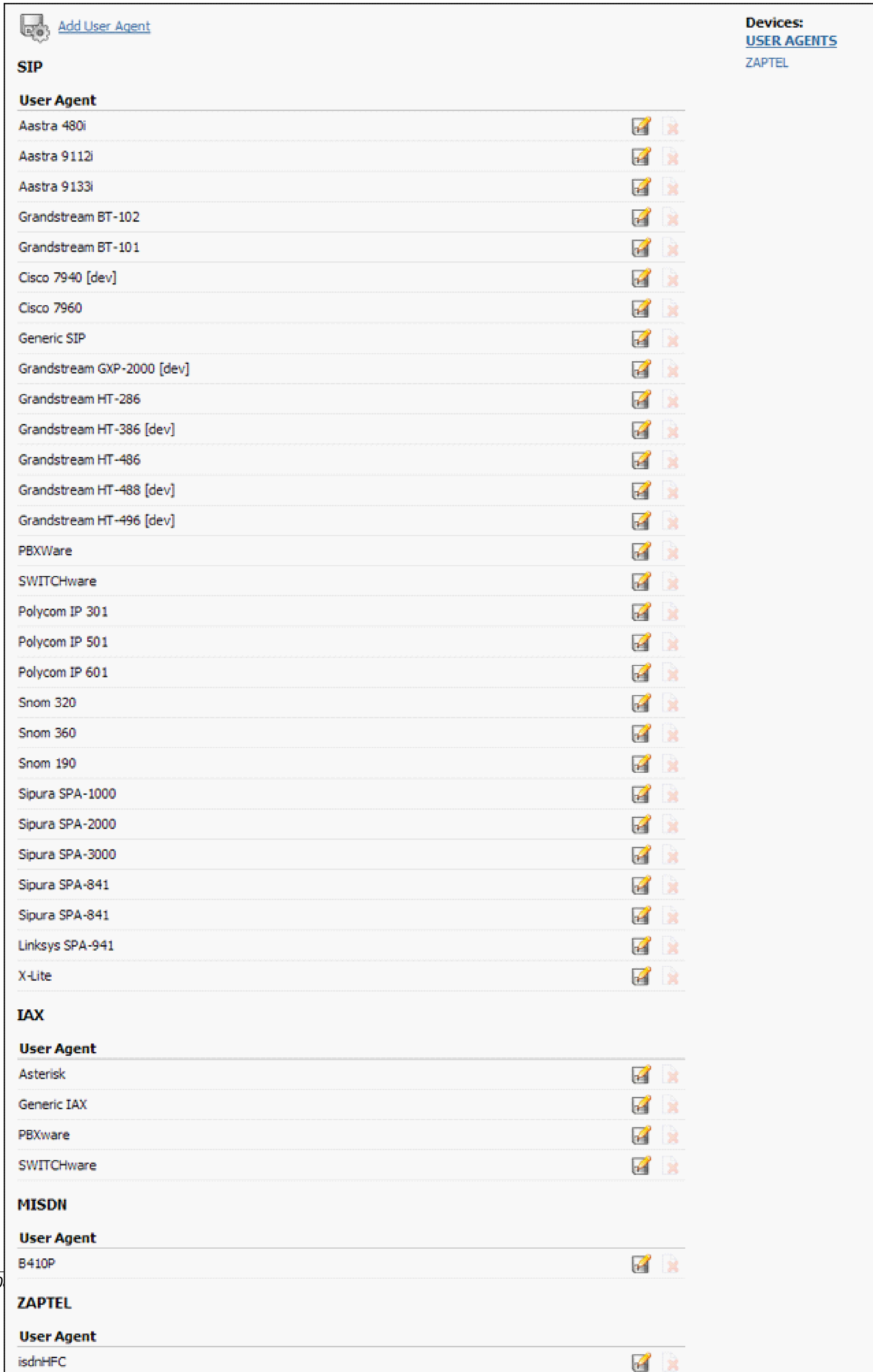
Settings:
- Default Trunks: All System calls go through trunks defined here
. MiniLCR: Overrides 'Default Trunks' and sets a specific trunk for a destination

Extensions:
- Trunks: Overrides 'Settings: Default Trunks'
- Routes: Overrides 'Settings: MiniLCR'

21.5 UAD

UAD (User Agent Devices) are various IP phones, soft phones, ATA (Analog Telephone Adaptors) and IAD (Integrated Access Devices) used for system extensions. PBXware supports a wide range of UAD using SIP,IAX, MGCP and ZAPTEL protocols.

Supported devices are already pre-configured with most common settings in order to allow administrators an easy way of adding extensions. However, some PBXware installation have a specific requirements hence it is advisable to edit selected UAD and set it to required values. Additionally if an installation needs o use an UAD not listed, clicking on "Add User Agent" allows adding new UAD.


Add User Agent
Devices:
USER AGENTS
ZAPTEL
SIP
User Agent
Aastra 480i
Aastra 9112i
Aastra 9133i
Grandstream BT-102
Grandstream BT-101
Cisco 7940 [dev]
Cisco 7960
Generic SIP
Grandstream GXP-2000 [dev]
Grandstream HT-286
Grandstream HT-386 [dev]
Grandstream HT-486
Grandstream HT-488 [dev]
Grandstream HT-496 [dev]
PBXWare
SWITCHware
Polycom IP 301
Polycom IP 501
Polycom IP 601
Snom 320
Snom 360
Snom 190
Sipura SPA-1000
Sipura SPA-2000
Sipura SPA-3000
Sipura SPA-841
Sipura SPA-841
Linksys SPA-941
X-Lite
IAX
User Agent
Asterisk
Generic IAX
PBXware
SWITCHware
MISDN
User Agent
B410P
ZAPTEL
User Agent
isdnHFC

21.5.1 Requirements

In this chapter we will cover:

21.5.1.1 Paging

Paging is a service that supports transmitting of messages to multiple phones over their loudspeakers

In this chapter we will cover:

21.5.1.1.1 Budgetone 101/102

1. Navigate Internet browser to phone IP address (for example http://192.168.1.1)

2. Provide admin password ('admin' by default)

3. Click 'Login' button

4. Click on 'Advanced Settings' link

5. Scroll down to 'Auto-Answer' options

6. Select 'Yes'

7. Click 'Update' button

8. Click 'Reboot' button

NOTE: Paging tested with Firmware version 1.0.8.16

21.5.1.1.2 GXP 2000

1. Navigate Internet browser to phone IP address (for example http://192.168.1.1)

2. Provide admin password ('admin' by default)

3. Click 'Login' button

4. Click on 'Account *' you wish to edit

5. Scroll down to 'Auto-Answer' options

6. Select 'Yes'

7. Click 'Update' button

8. Click 'Reboot' button

NOTE: Paging tested with Firmware version 1.0.1.12

21.5.1.1.3 Snom 190/320

1. Navigate Internet browser to phone IP address (for example http://192.168.1.1)

2. Navigate to your line e.g. 'Line 1'

3. Click 'SIP'

4. Set 'Auto Answer' to 'On'

5. Click 'Save' button

6. Navigate to 'Preferences'

7. Set 'Auto Answer Indication' to 'On' for a sound to be played notifying you a call has been received

8. Set 'Type of Answering' to suit your needs e.g. 'Handsfree'

9 Click 'Save' button

NOTE: Paging tested with Firmware version 5.2b

21.5.1.1.4 Polycom 30x/50x/60x

You need the latest version of both the SIP software and bootROM to do it. Auto-answer could be configured only using provisioning. To prepare configuration files you have to do following steps:

1. In the 'sip.cfg' file, look for the line with these variables:
<alertInfo voIpProt.SIP.alertinfo.1.value="Auto Answer" voIpProt.SIP.alertInfo.1.class="3"...>
Polycom calls up class 3 in sip.cfg or ipmid.cfg file.

2. In 'sip.cfg' my ring class 'AUTO_ANSWER' looks like this:
<ringType se.rt.enabled="1" se.rt.modification.enabled="1">
<DEFAULT se.rt.1.name="Default" se.rt.1.type="ring" se.rt.1.ringer="2" se.rt.1.callWait="6" se.rt.1.mod="1"/>
<VISUAL_ONLY se.rt.2.name="Visual" se.rt.2.type="visual"/>
<AUTO_ANSWER se.rt.3.name="Auto Answer" se.rt.3.type="answer"/>
<RING_ANSWER se.rt.4.name="Ring Answer" se.rt.4.type="ring-answer" se.rt.4.timeout="2000" se.rt.4.ringer="2" se.rt.4.callWait="6" se.rt.4.mod="1"/>
<INTERNAL se.rt.5.name="Internal" se.rt.5.type="ring" se.rt.5.ringer="2" se.rt.5.callWait="6" se.rt.5.mod="1"/>
<EXTERNAL se.rt.6.name="External" se.rt.6.type="ring" se.rt.6.ringer="2" se.rt.6.callWait="6" se.rt.6.mod="1"/>
<EMERGENCY se.rt.7.name="Emergency" se.rt.7.type="ring" se.rt.7.ringer="2" se.rt.7.callWait="6" se.

rt.7.mod="1"/>
<CUSTOM_1 se.rt.8.name="Custom 1" se.rt.8.type="ring" se.rt.8.ringer="5" se.rt.8.callWait="7" se.rt.8.mod="1"/>
<CUSTOM_2 se.rt.9.name="Custom 2" se.rt.9.type="ring" se.rt.9.ringer="7" se.rt.9.callWait="7" se.rt.9.mod="1"/>
<CUSTOM_3 se.rt.10.name="Custom 3" se.rt.10.type="ring" se.rt.10.ringer="9" se.rt.10.callWait="7" se.rt.10.mod="1"/>
<CUSTOM_4 se.rt.11.name="Custom 4" se.rt.11.type="ring" se.rt.11.ringer="11" se.rt.11.callWait="7" se.rt.11.mod="1"/>
</ringType>

'se.rt.3.type="ANSWER"' sets Polycom phone ring type, in this case an answer, that means that phone will automatically answer without ringing.

3. Update modified files to provisioning server

4. Reload PBXware if used as provisioning server

5. Restart phone

6.

The bootROM on the phone performs the provisioning functions of downloading the bootROM, the <Ethernet address>.cfg file, and the SIP application and uploading log files. The SIP application performs the provisioning functions of downloading all other configuration files, uploading and downloading the configuration override file and user directory, downloading the dictionary and uploading log files.

The protocol which will be used to transfer files from the boot server depends on sev-eral factors including the phone model and whether the bootROM or SIP application stage of provisioning is in progress. TFTP and FTP are supported by all SoundPoint® and SoundStation® phones. The SoundPoint® IP 301, 501, 600 and 601 and Sound- Station® IP 4000 bootROM also supports HTTP while the SIP application supports HTTP1 and HTTPS. If an unsupported protocol is specified, this may result in unex-pected behavior, see the table for details of which protocol the phone will use. The "Specified Protocol" listed in the table can be selected in the Server Type field or the Server Address can include a transfer protocol, for example http://usr:pwd@server (see 2.2.1.3.3 Server Menu on page 10). The boot server address can also be obtained via DHCP. Configuration file names in the <Ethernet address>.cfg file can include a transfer protocol, for example https://usr:pwd@server/dir/file.cfg. If a user name and password are specified as part of the server address or file name, they will be used only if the server supports them.

NOTE: A URL should contain forward slashes instead of back slashes and should not contain spaces. Escape characters are not supported. If a user name and password are not specified, the Server User and Server Password will be used.

Specified Protocol	Protocol used by bootROM		Protocol used by SIP Application	
	300, 500	301, 501, 600, 601, 4000	300, 500	301, 501, 600, 601, 4000
FTP	FTP	FTP	FTP	FTP
TFTP	TFTP	TFTP	TFTP	TFTP
HTTP	**FTP**	HTTP	HTTP	HTTP
HTTPS	**FTP**	**HTTP**	Not supported. Transfers will fail.	HTTPS

For downloading the bootROM and application images to the phone, the secure HTTPS protocol is not available. To guarantee software integrity, the bootROM will only download signed bootROM or application images. For HTTPS, widely recog-nized certificate authorities are trusted by the phone and custom certificates can be added. See 6.1 Trusted Certificate Authority List on page 151. Using HTTPS requires that SNTP be functional. Provisioning of configuration files is done by the application instead of the bootROM and this transfer can use a secure protocol.

Configuring Cisco phones for speakerphone page.
Note that Cisco phones are not supporting special sip header for auto-answer, that means that You can’t configure these phones to auto answer only to speaker phone page and to ring for ordinary incoming calls. You can only configure one line who will be set to auto answer any call not just page call. That means that with Cisco you have to have one special separated line for speakerphone page.

To set already configured line on Cisco phone you have to do following steps:

1. Go into menu 'Settings -> Call Preferences -> Auto Answer (intercom)'

2. Set a new line you've just created as auto-answer by choosing 'Line x ON/OFF'

NOTE: Paging tested with Firmware version 1.6.5.0043

21.5.1.1.5 Cisco 7940-7960

1. Create a new line on a Cisco phone, and put the configuration into sip.conf as you normally would(go into 'Settings: Call Preferences: Auto Answer (intercom)' and then make the line you've just created as 'auto-answer'.

2. Here are the contents of /var/lib/asterisk/agi-bin/callall:

```
#!/bin/sh
cp  /var/lib/asterisk/agi-bin/*conf  /var/spool/asterisk/outgoing
```

3. Make sure to make the script executable. And then for every extension you have as an auto-answer, have a file like this
in /var/lib/asterisk/agi-bin:

```
Channel:  SIP/2006
Context:   add-to-conference
WaitTime:  2
Extension:  start
Priority: 1
CallerID: Office Pager <5555>
```

So, for example, if you have three lines that are configured for automatic answering - SIP/2006, SIP/2007, SIP/2008, you should have three files named 2006-conf, 2007-conf, 2008-conf in /var/lib/asterisk/agi-bin that get copied into the outgoing call spool directory every time you call extension 5555.

4. Now, dial 5555 from any phone and you should have one-way paging.

People who use the pager may have to get used to waiting 1-2 seconds before speaking to allow all the phones to catch up with the audio stream. All of the phones hang up after 20 seconds, regardless of if the person originating the page has stopped talking. Change the AbsoluteTimeout values to increase this interval.

If you want a really confusing loud mess, then change the "dmq" options to "dq" and you'll get an N-way conversation going with everyone who has a phone. Bad.

If you want a really interesting office surveillance tool, change the "dmq" to "dt" and you'll suddenly be listening to all of the extensions in the office, like some kind of mega-snoop tool. Useful for after-hours listening throughout the entire office.

NOTE: Paging tested with Firmware version 6.1

21.5.1.1.6 Linksys 941

1. Navigate Internet browser to phone IP address (for example http://192.168.1.1)

2. Select 'Admin Login'

3. Select 'Advanced'

4. Navigate to 'User' tab

5. Set 'Auto Answer Page' to 'Yes'

6. Set 'Send Audio To Speaker' to 'Yes'

7. Click 'Submit All Changes' button

NOTE: At the time paging option is set for all lines and works once handset is picked up. Paging tested with Firmware version 4.1.12(a)

21.5.1.1.7 Aastra 480i/9133i/9112i

1. Navigate Internet browser to phone IP address (for example http://192.168.1.1)

2. Select 'Preferences' under 'Basic Settings' tab

3. Set 'Microphone Mute' to 'No'

4. Set 'Auto-Answer' to 'Yes'

5. Click 'Save Settings' button

6. Reboot the phone to apply the new settings

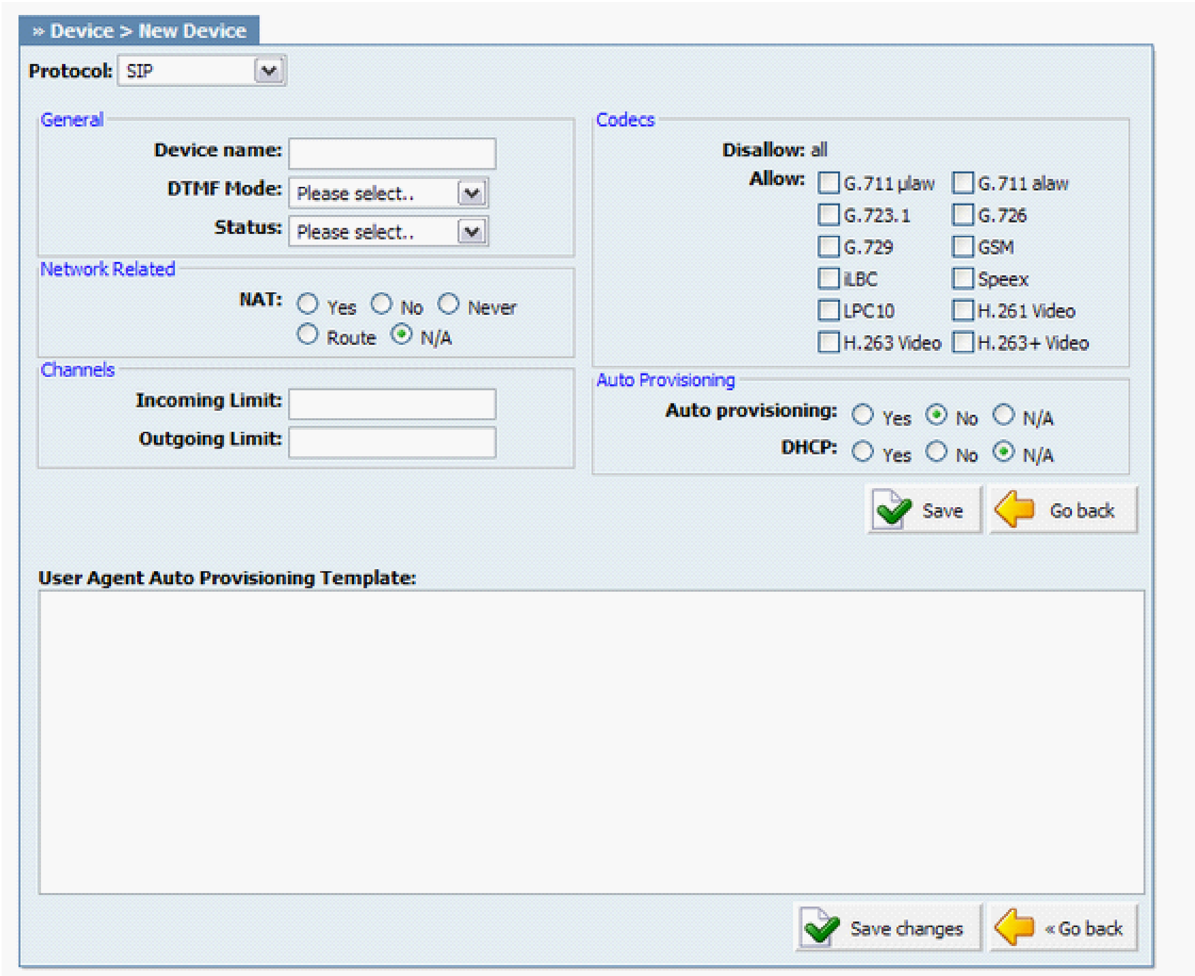
NOTE: At the time paging option is set for all lines and works once handset is picked up. Paging tested with Firmware version 1.3.1.1095

21.5.2 Add/Edit

In this chapter we will cover:
- SIP 423
- IAX 426
- MGCP 431
- ZAPTEL 435

21.5.2.1 SIP

Click 'Add User Agent' to add a device or click 'Edit' icon next to one, to change its settings.



Protocol:

Select protocol UAD(User Agent Device) uses

Example: If UAD/Phone uses SIP protocol, select 'SIP' here
Field Type: Select box

Device Name:

Unique device name

Example: AASTRA 480i
Field Type: [a-z][0-9]

DTMF Mode (Dual Tone Multi-Frequency) **:**

A specific frequency, consisting of two separate tones. Each key has a specific tone assign to it so it can be easily identified by a microprocessor.

Example: This is a sound heard when dialing digits on touch-tone phones. Each phone has different 'DTMF Mode'. By default, this field is populated automatically for supported devices. If adding other UAD/Phone select between 'inband', **,** 'rfc2833' or 'info' options
Field Type: Select box

Status:

Extension status/presence on the network

Example: If this field is set to 'Active', this UAD/Phone will be available in the 'UAD' select box by default when adding new extensions
Field Type: Select Box

Status:

Extension status/presence on the network

Example: If this field is set to 'Active', this UAD/Phone will be available in the 'UAD' select box by default when adding new extensions
Field Type: Select Box

NAT (Network Address Translation):

Set the appropriate Extension - PBXware NAT relation

Example:
If Extension 1000 is trying to register with the PBXware from a remote location/network and that network is behind NAT, select the appropriate NAT settings here:

- **yes** - Always ignore info and assume NAT
- **no** - Use NAT mode only according to RFC3581
- **never** - Never attempt NAT mode or RFC3581 support
- **route** - Assume NAT, don't send rport

Field Type: Option buttons

Incoming Dial Options:

Advanced dial options for all incoming calls

Example: Please see below for detail list of all available dial options(default: tr)
Field Type: [a-z]

Outgoing Dial Options:

Advanced dial options for all outgoing calls

Example: Please see below for detail list of all available dial options(default: empty)
Field Type: [a-z]

Disallow:

Set the codecs extension is now allowed to use

Example: This field is very unique. In order to work properly, this setting is automatically set to 'Disallow All' and it cannot be modified
Field Type: Read only

Allow:

Set the codecs extension is allowed to use

Example: Only the codecs set under 'Settings: Server' will be available to choose from
Field Type: Check box

- **ITU G.711 ulaw** **-** 64 Kbps, sample-based, used in US
- **ITU G.711 alaw** - 64 Kbps, sample-based, used in Europe
- **ITU G.723.1** - (5.3/6.3 Kbps, 30ms frame size
- **ITU G.726** - 16/24/32/40 Kbps
- **ITU G.729** - 8 Kbps, 10ms frame size
- **GSM** - 13 Kbps (full rate), 20ms frame size
- **iLBC** - 15Kbps,20ms frame size: 13.3 Kbps, 30ms frame size
- **Speex** - 2.15 to 44.2 Kbps
- **LPC10** - 2.5 Kbps
- **H.261 Video** - Used over ISDN lines with resolution of 352x288
- **H.263 Video** - Low-bit rate encoding solution for video conferencing
- **H.263+ Video** - Extension of H.263 that provides additional features that improve compression over packet switched networks.

Auto Provisioning:

Enable auto provisioning service for this extension

Example: Connect UAD/Phone to PBXware without any hassle by providing UAD/Phone MAC address(and optionally adding Static UAD/Phone IP address and network details)
Field Type: Option Buttons

DHCP (Dynamic Hosts Configuration Protocol):

Set whether UAD/Phone is on DHCP or Static IP address

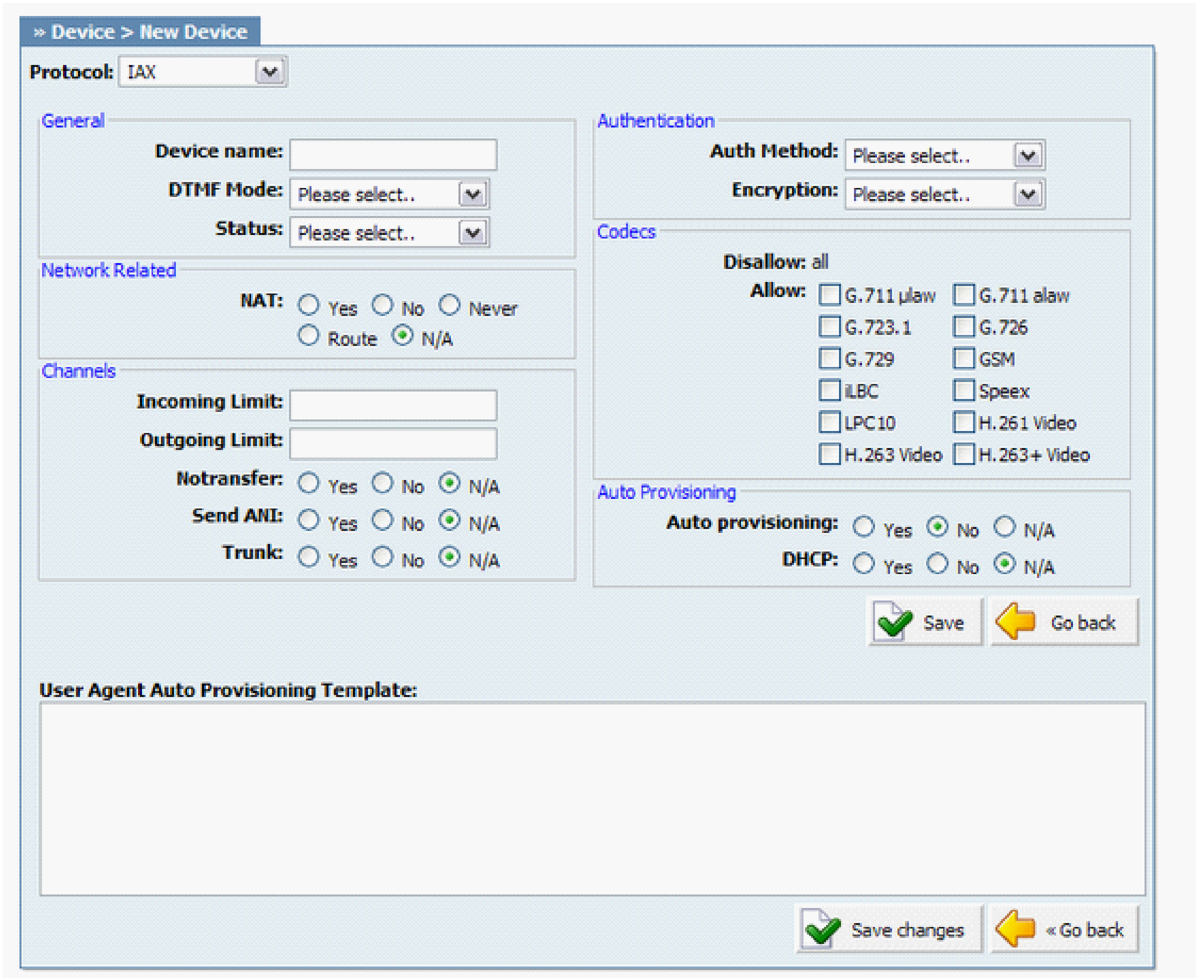
Example: Set DHCP = Yes if UAD/Phone is on dynamic or DHCP = No if UAD/Phone is on static IP address.
If on static IP, you will have to provide more network details in the fields bellow.
Field Type: Option buttons

User Agent Auto Provisioning Template:

Example: This option is used for providing additional config parameters for SIP, IAX and MGCP configuration files.
Values provided here will be written into these configuration files.
Field Type: [a-z][0-9]

21.5.2.2 IAX

Click 'Add User Agent' to add a device or click 'Edit' icon next to one, to change its settings.



Protocol:

Select protocol UAD(User Agent Device) uses

Example: If UAD/Phone uses SIP protocol, select 'SIP' here
Field Type: Select box

Device Name:

Unique device name

Example: AASTRA 480i
Field Type: [a-z][0-9]

DTMF Mode (Dual Tone Multi-Frequency) **:**

A specific frequency, consisting of two separate tones. Each key has a specific tone assign to it so it can be easily identified by a microprocessor.

Example: This is a sound heard when dialing digits on touch-tone phones. Each phone has different 'DTMF Mode'. By default, this field is populated automatically for supported devices. If adding other UAD/Phone select between 'inband', **,** 'rfc2833' or 'info' options
Field Type: Select box

Status:

Extension status/presence on the network

Example: If this field is set to 'Active', this UAD/Phone will be available in the 'UAD' select box by default when adding new extensions
Field Type: Select Box

NAT (Network Address Translation):

Set the appropriate Extension - PBXware NAT relation

Example:
If Extension 1000 is trying to register with the PBXware from a remote location/network and that network is behind NAT, select the appropriate NAT settings here:

- **yes** - Always ignore info and assume NAT
- **no** - Use NAT mode only according to RFC3581
- **never** - Never attempt NAT mode or RFC3581 support
- **route** - Assume NAT, don't send rport

Field Type: Option buttons

Incoming Dial Options:

Advanced dial options for all incoming calls

Example: Please see below for detail list of all available dial options(default: tr)
Field Type: [a-z]

Outgoing Dial Options:

Advanced dial options for all outgoing calls

Example: Please see below for detail list of all available dial options(default: empty)
Field Type: [a-z]

Notransfer:

Disable native IAX transfer

Example:
Field Type: Option buttons

Send ANI:

Should ANI ("super" Caller ID) be sent over this Provider

Example: Set 'Yes' to enable
Field Type: Option buttons

Trunk:

Use IAX2 trunking with this host

Example: Set 'Yes' to enable
Field Type: Option buttons

Auth Method:

Authentication method required by provider

Example: md5
Field Type: [a-z] [0-9]

Encryption:

Should encryption be used when authenticating with the peer

Example:
Field Type: [a-z][0-9]

Disallow:

Set the codecs extension is now allowed to use

Example: This field is very unique. In order to work properly, this setting is automatically set to 'Disallow All' and it cannot be modified
Field Type: Read only

Allow:

Set the codecs extension is allowed to use

Example: Only the codecs set under 'Settings: Server' will be available to choose from
Field Type: Check box

- **ITU G.711 ulaw** - 64 Kbps, sample-based, used in US
- **ITU G.711 alaw** - 64 Kbps, sample-based, used in Europe
- **ITU G.723.1** - (5.3/6.3 Kbps, 30ms frame size
- **ITU G.726** - 16/24/32/40 Kbps
- **ITU G.729** - 8 Kbps, 10ms frame size
- **GSM** - 13 Kbps (full rate), 20ms frame size
- **iLBC** - 15Kbps,20ms frame size: 13.3 Kbps, 30ms frame size
- **Speex** - 2.15 to 44.2 Kbps
- **LPC10** - 2.5 Kbps
- **H.261 Video** - Used over ISDN lines with resolution of 352x288
- **H.263 Video** - Low-bit rate encoding solution for video conferencing
- **H.263+ Video** - Extension of H.263 that provides additional features that improve compression over packet switched networks.

Auto Provisioning:

Enable auto provisioning service for this extension

Example: Connect UAD/Phone to PBXware without any hassle by providing UAD/Phone MAC address(and optionally adding Static UAD/Phone IP address and network details)
Field Type: Option Buttons

DHCP (Dynamic Hosts Configuration Protocol):

Set whether UAD/Phone is on DHCP or Static IP address

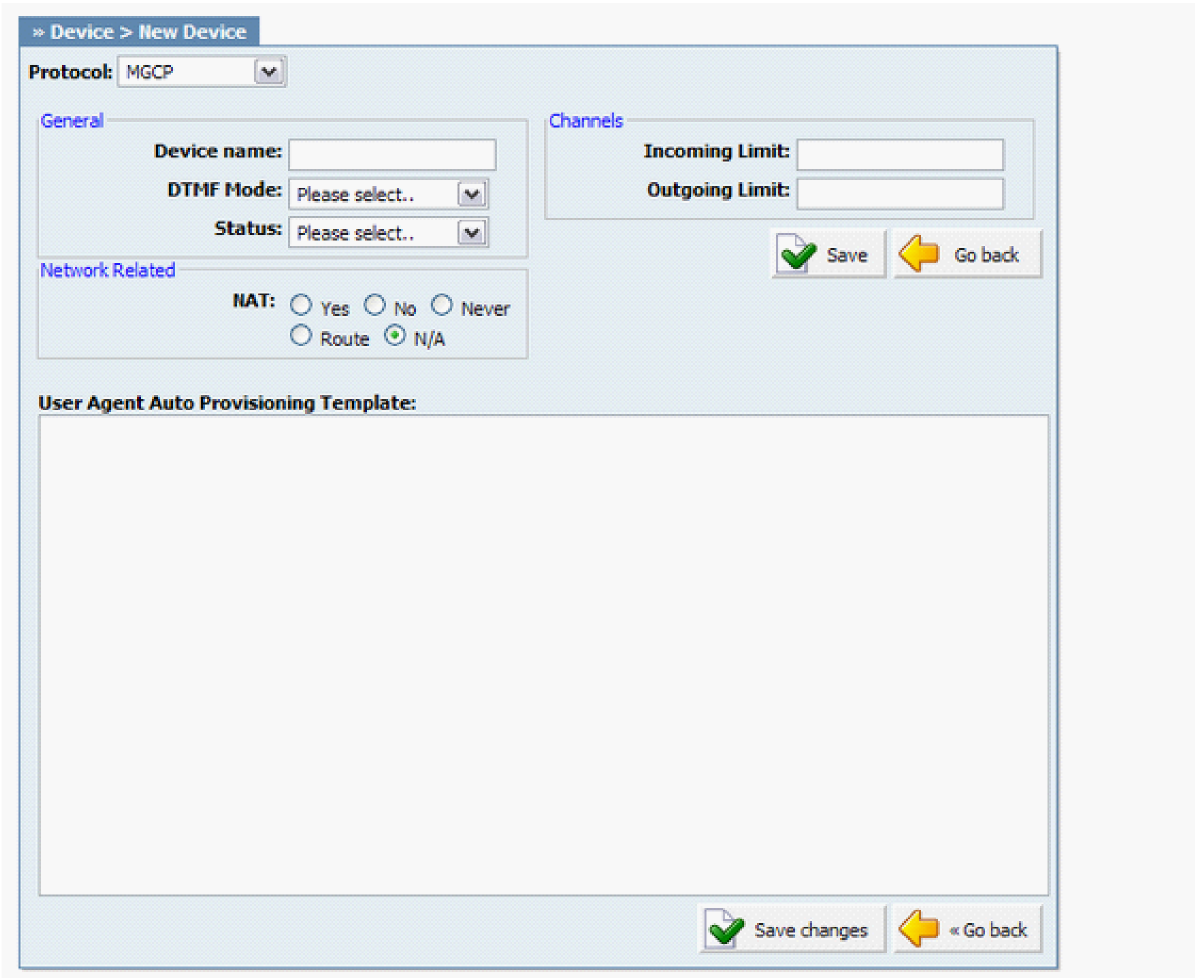
Example: Set DHCP = Yes if UAD/Phone is on dynamic or DHCP = No if UAD/Phone is on static IP address. If on static IP, you will have to provide more network details in the fields bellow.
Field Type: Option buttons

User Agent Auto Provisioning Template:

Example: This option is used for providing additional config parameters for SIP, IAX and MGCP configuration files.
Values provided here will be written into these configuration files.
Field Type: [a-z][0-9]

21.5.2.3 MGCP

Click 'Add User Agent' to add a device or click 'Edit' icon next to one, to change its settings.



Protocol:

Select protocol UAD(User Agent Device) uses

Example: If UAD/Phone uses SIP protocol, select 'SIP' here
Field Type: Select box

Device Name:

Unique device name

Example: AASTRA 480i

Field Type: [a-z][0-9]

DTMF Mode (Dual Tone Multi-Frequency) **:**

A specific frequency, consisting of two separate tones. Each key has a specific tone assign to it so it can be easily identified by a microprocessor.

Example: This is a sound heard when dialing digits on touch-tone phones. Each phone has different 'DTMF Mode'. By default, this field is populated automatically for supported devices. If adding other UAD/Phone select between 'inband', **,** 'rfc2833' or 'info' options
Field Type: Select box

Status:

Extension status/presence on the network

Example: If this field is set to 'Active', this UAD/Phone will be available in the 'UAD' select box by default when adding new extensions
Field Type: Select Box

NAT (Network Address Translation):

Set the appropriate Extension - PBXware NAT relation

Example:
If Extension 1000 is trying to register with the PBXware from a remote location/network and that network is behind NAT, select the appropriate NAT settings here:

- **yes** - Always ignore info and assume NAT
- **no** - Use NAT mode only according to RFC3581
- **never** - Never attempt NAT mode or RFC3581 support
- **route** - Assume NAT, don't send rport

Field Type: Option buttons

Incoming Dial Options:

Advanced dial options for all incoming calls

Example: Please see below for detail list of all available dial options(default: tr)
Field Type: [a-z]

Outgoing Dial Options:

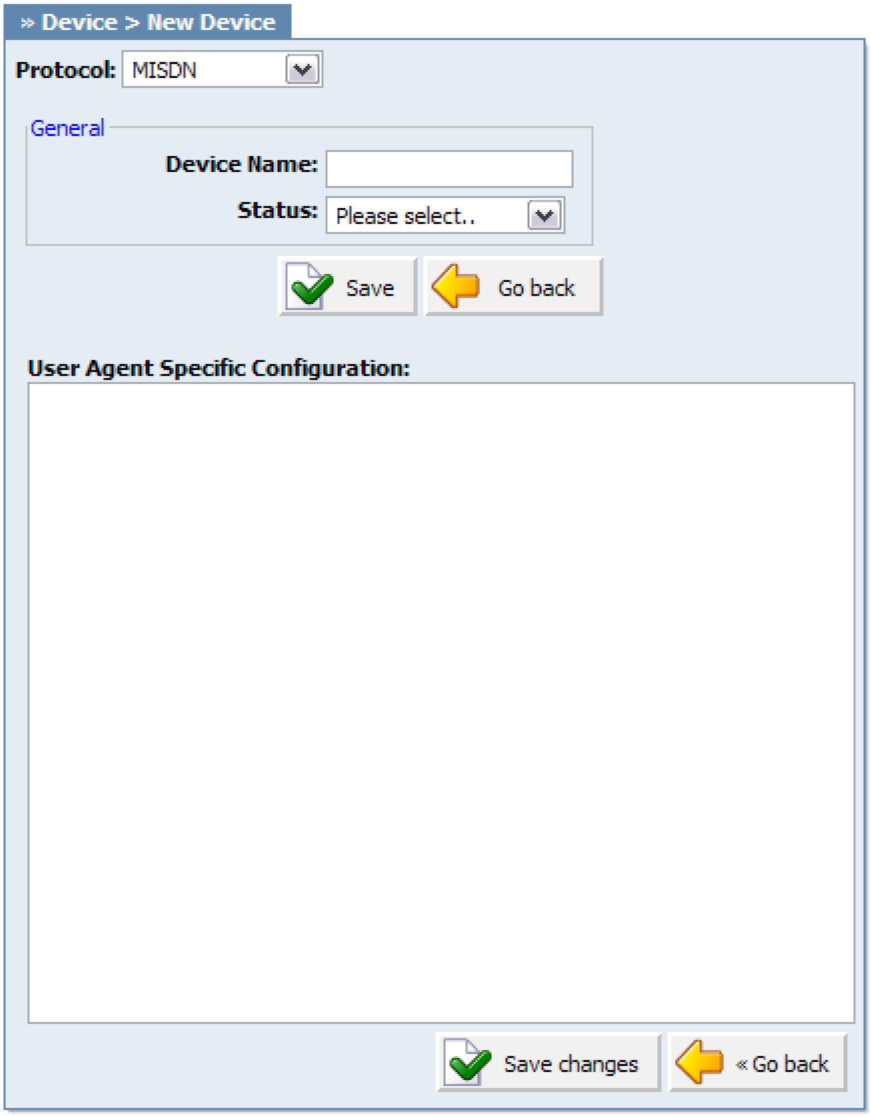
Advanced dial options for all outgoing calls

Example: Please see below for detail list of all available dial options(default: empty)
Field Type: [a-z]

User Agent Auto Provisioning Template:

Example: This option is used for providing additional config parameters for SIP, IAX and MGCP configuration files.
Values provided here will be written into these configuration files.
Field Type: [a-z][0-9]

21.5.2.4 MISDN



Protocol:

Select protocol UAD(User Agent Device) uses

Example: If UAD/Phone uses SIP protocol, select 'SIP' here
Field Type: Select box

Device Name:

Unique device name

Example: AASTRA 480i
Field Type: [a-z][0-9]

Status:

Extension status/presence on the network

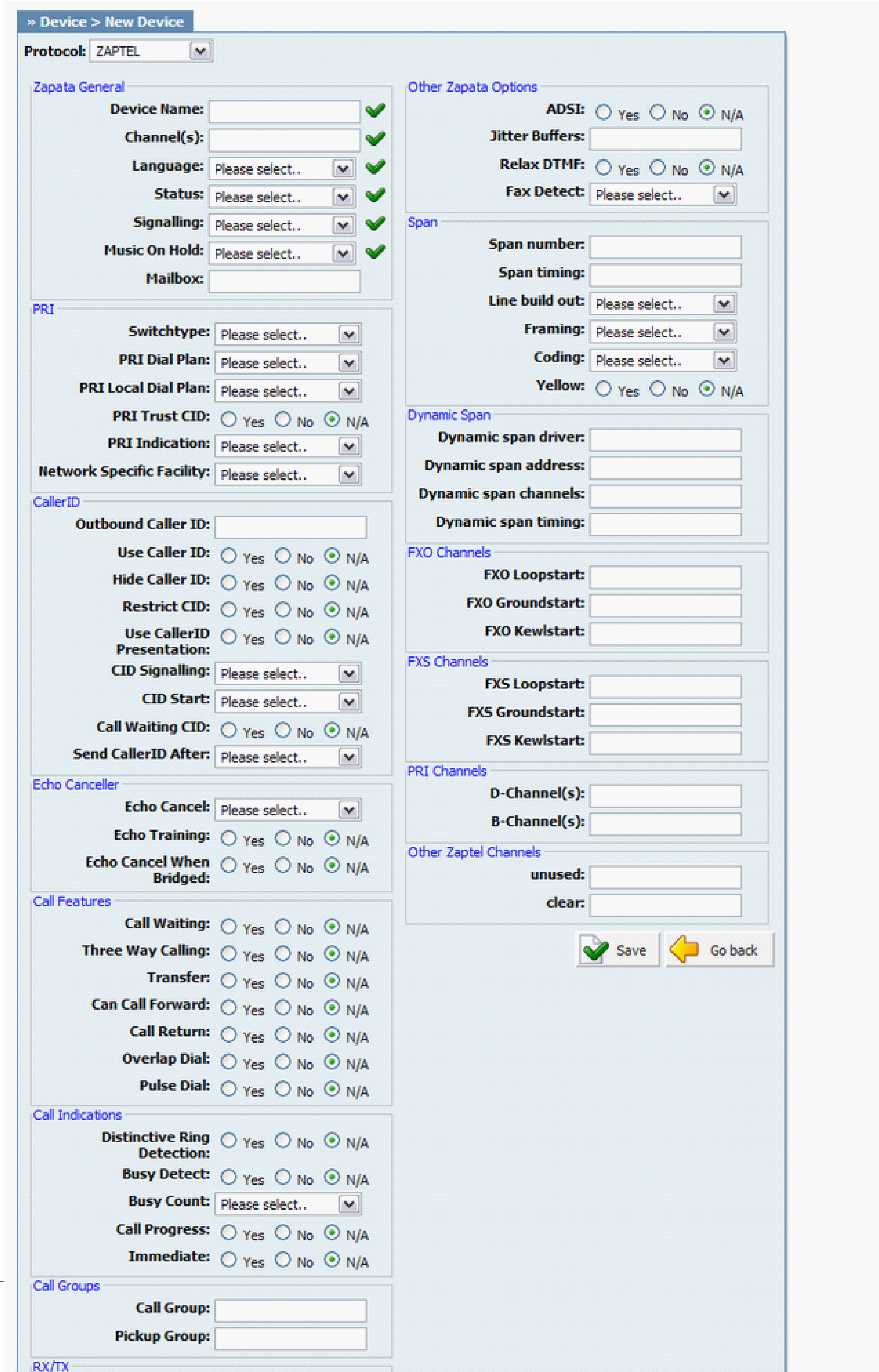
Example: If this field is set to 'Active', this UAD/Phone will be available in the 'UAD' select box by default when adding new extensions
Field Type: Select Box

User Agent Auto Provisioning Template:

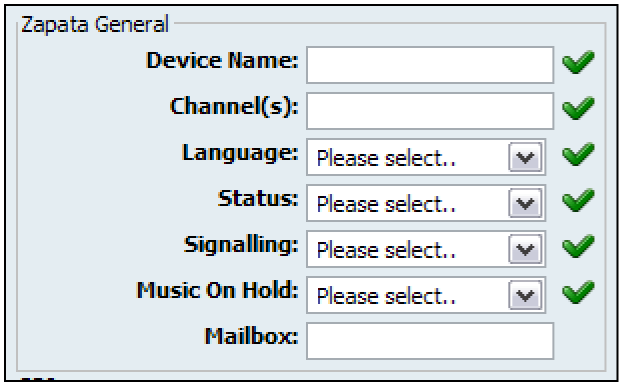
Example: This option is used for providing additional config parameters for SIP, IAX and MGCP configuration files.
Values provided here will be written into these configuration files.
Field Type: [a-z][0-9]

21.5.2.5 ZAPTEL

In this chapter we will cover:


» Device > New Device
Protocol: ZAPTEL
Zapata General
Device Name:
Channel(s):
Language: Please select..
Status: Please select..
Signalling: Please select..
Music On Hold: Please select..
Mailbox:
PRI
Switchtype: Please select..
PRI Dial Plan: Please select..
PRI Local Dial Plan: Please select..
PRI Trust CID: Yes No N/A
PRI Indication: Please select..
Network Specific Facility: Please select..
CallerID
Outbound Caller ID:
Use Caller ID: Yes No N/A
Hide Caller ID: Yes No N/A
Restrict CID: Yes No N/A
Use CallerID Presentation: Yes No N/A
CID Signalling: Please select..
CID Start: Please select..
Call Waiting CID: Yes No N/A
Send CallerID After: Please select..
Echo Canceller
Echo Cancel: Please select..
Echo Training: Yes No N/A
Echo Cancel When Bridged: Yes No N/A
Call Features
Call Waiting: Yes No N/A
Three Way Calling: Yes No N/A
Transfer: Yes No N/A
Can Call Forward: Yes No N/A
Call Return: Yes No N/A
Overlap Dial: Yes No N/A
Pulse Dial: Yes No N/A
Call Indications
Distinctive Ring Detection: Yes No N/A
Busy Detect: Yes No N/A
Busy Count: Please select..
Call Progress: Yes No N/A
Immediate: Yes No N/A
Call Groups
Call Group:
Pickup Group:
RX/TX
Other Zapata Options
ADSI: Yes No N/A
Jitter Buffers:
Relax DTMF: Yes No N/A
Fax Detect: Please select..
Span
Span number:
Span timing:
Line build out: Please select..
Framing: Please select..
Coding: Please select..
Yellow: Yes No N/A
Dynamic Span
Dynamic span driver:
Dynamic span address:
Dynamic span channels:
Dynamic span timing:
FXO Channels
FXO Loopstart:
FXO Groundstart:
FXO Kewlstart:
FXS Channels
FXS Loopstart:
FXS Groundstart:
FXS Kewlstart:
PRI Channels
D-Channel(s):
B-Channel(s):
Other Zaptel Channels
unused:
clear:
Save
Go back

21.5.2.5.1 Zapata General



Device Name:

Unique device name

Example: AASTRA 480i
Field Type: [a-z][0-9]

Channels

Which card channels are used

Example: 1,4/1-4
Field Type: [0-9][,-]

Language:

Default language

Example: us
Field Type: Select box

Status:

Extension status/presence on the network

Example: If this field is set to 'Active', this UAD/Phone will be available in the 'UAD' select box by default when adding new extensions
Field Type: Select Box

Signalling:

Signalling method

Example: default

- FXS Loopstart
- FXS Groundstart
- FXS Kewlstart
- FXO Loopstart
- FXO Groundstart
- FXO Kewlstart
- PRI CPE side
- PRI Network side
- BRI CPE side
- BRI Network side
- BRI CPE PTMP
- BRI Network PTMP

Field Type: Select box

Music On Hold:

Select which class of music to use for music on hold. If not specified then the 'default' will be used

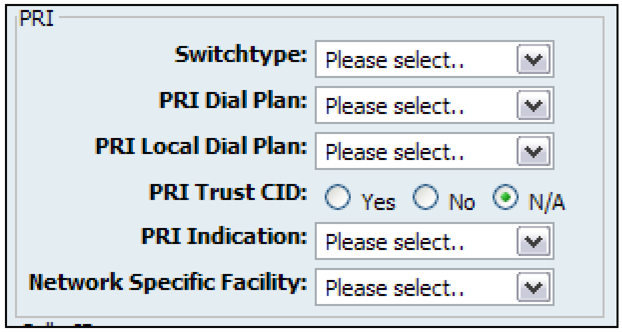
Example: default
Field Type: Select box

Mailbox:

Define a voicemail context

Example: 1234, 1234@context
Field Type: [a-z][0-9]

21.5.2.5.2 PRI



Switchtype:

Set switch type

Example:
- National ISDN 2
- Nortel DMS100
- AT&T 4ESS
- Lucent 5ESS
- EuroISDN
- Old National ISDN 1

Field Type: Select box

PRI Dial Plan:

Set dial plan used by some switches

Example:
- Unknown
- Private ISDN
- Local ISDN
- National ISDN
- International ISDN

Field Type: Select box

PRI Local Dial Plan:

Set numbering dial plan for destinations called locally

Example:
- Unknown

- Private ISDN
- Local ISDN
- National ISDN
- International ISDN

Field Type: Select box

PRI Trust CID:

Trust provided caller id information

Example: Yes, No, N/A
Field Type: Option buttons

PRI Indication:

How to report 'busy' and 'congestion' on a PRI

Example:
- **outofband -** Signal Busy/Congestion out of band with RELEASE/DISCONNECT
- **inband -** Signal Busy/Congestion using in-band tones

Field Type: Select box

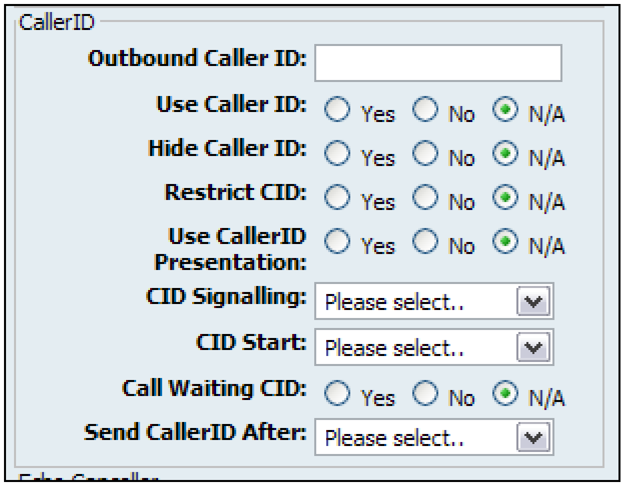
Network Specific Facility:

If required by switch, select network specific facility

Example:
- none
- sdn
- megacom
- accunet

Field Type: Select box

21.5.2.5.3 Caller ID



Outbound Caller ID:

Caller ID set for all outbound calls where Caller ID is not set or supported by a device

Example: john@domain.com
Field Type: [0-9]

Caller ID:

CallerID can be set to 'asreceived' or a specific number if you want to override it

Example: 'asreceived', 555648788
Field Type: [a-z][0-9]

NOTE: Caller ID can only be transmitted to the public phone network with supported hardware, such as a PRI. It is not possible to set external caller ID on analog lines

Use Caller ID:

Whether or not to use caller id

Example: Yes, No, N/A
Field Type: Options buttons

Hide Caller ID:

Whether or not to hide outgoing caller ID

Example: Yes, No, N/A
Field Type: Options buttons

Restrict CID

Whether or not to use the caller ID presentation for the outgoing call that the calling switch is sending

Example: Yes, No, N/A
Field Type: Options buttons

CID Signalling:

Set the type of caller ID signalling

Example:
- **bell** - US
- **v23** - UK
- **dtmf** - Denmark, Sweden and Netherlands

Field Type: Select box

CID Start:

What signals the start of caller ID

Example:
- ring = a ring signals the start
- polarity = polarity reversal signals the start

Field Type: Select box

Call Waiting CID:

Whether or not to enable call waiting on FXO lines

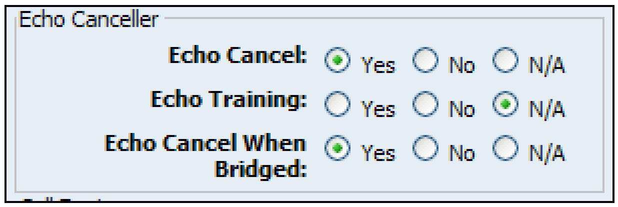
Example: Yes, No, N/A
Field Type: Options buttons

Send CallerID After:

Some countries, like UK, have different ring tones (ring-ring), which means the caller id needs to be set later on, and not just after the first ring, as per the default.

Example: Yes
Field Type: Select box

21.5.2.5.4 Echo Canceller



Echo Cancel:

Enable echo cancellation

Example: Yes, No, N/A
Field Type: Option buttons

Echo Training:

Mute the channel briefly, for 400ms, at the beginning of conversation, cancelling the echo. (Use this only if 'Echo Cancel' doesn't work as expected)

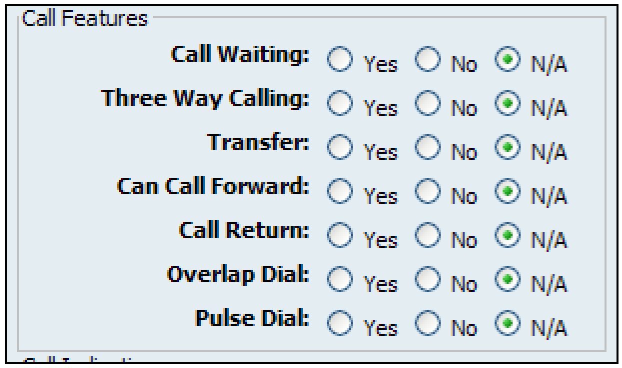
Example: Yes, No, N/A
Field Type: Option buttons

Echo Cancel When Bridged:

Enable echo cancellation when bridged. Generally not necessary, and in fact undesirable, to echo cancel when the circuit path is entirely TDM

Example: Yes, No, N/A
Field Type: Option buttons

21.5.2.5.5 Call Features



Call Waiting:

Whether or not to enable call waiting on FXO lines

Example: Yes, No, N/A
Field Type: Option buttons

Use Calling Pres:

Whether or not use the caller ID presentation for the outgoing call that the calling switch is sending
Example: Yes, No, N/A

Field Type: Option buttons

Three Way Calling:

Support three-way calling. If enabled, call can be put on hold and one is able to make another call

Example: Yes, No, N/A
Field Type: Option buttons

Transfer:

Support call transfer and also enables call parking (overrides the 'canpark' parameter). Requires 'Three Way Calling' = 'Yes'.

Example: Yes, No, N/A
Field Type: Option buttons

Can Call Forward:

Support call forwarding

Example: Yes, No, N/A
Field Type: Option buttons

Call Return:

Whether or not to support Call Return **'*69'.** Dials last caller extension number

Example: Yes, No, N/A
Field Type: Option buttons

Overlap Dial:

Enable overlap dialing mode (sends overlap digits)

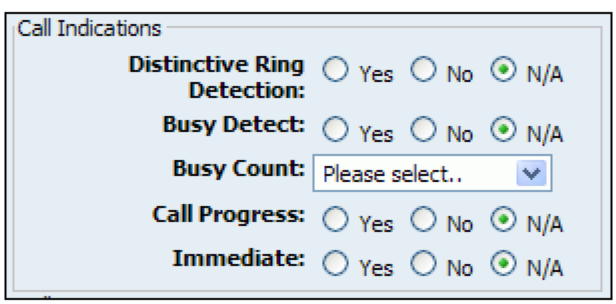
Example: Yes, No, N/A
Field Type: Option buttons

Pulse Dial:

Use pulse dial instead of DTMF. Used by FXO (FXS signalling) devices

Example: Yes, No, N/A
Field Type: Option buttons

21.5.2.5.6 Call Indications



Distinctive Ring Detection:

Whether or not to do distinctive ring detection on FXO lines

Example: Yes, No, N/A
Field Type: Options buttons

Busy Detect:

Enable listening for the beep-beep busy pattern

Example: Yes, No, N/A
Field Type: Options buttons

Busy Count:

How many busy tones to wait before hanging up. Bigger settings lower probability of random hangups. 'Busy Detect' has to be enabled

Example:
- 4
- 6
- 8

Field Type: Select box

Call Progress:

Easily detect false hangups

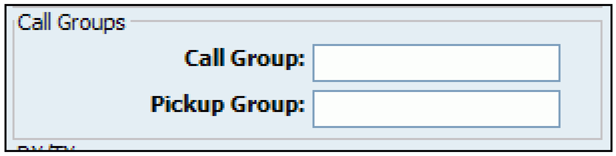
Example: Yes, No, N/A
Field Type: Options buttons

Immediate:

Should channel be answered immediately or the simple switch should provide dialtone, read digits, etc

Example: Yes, No, N/A
Field Type: Options buttons

21.5.2.5.7 Call Groups



Call Group:

Set the Call Group extension belongs to.

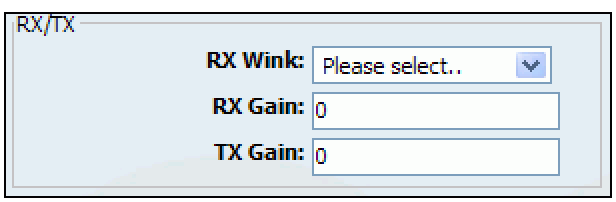
Example: Similar to 'Context' grouping, only this option sets to which call group extension belongs(Allowed range 0-63)
Field Type: [0-9] [,-]

Pickup Group:

Set groups extension is allowed to pickup.

Example: Similar to 'Context' grouping, only this option sets the Call Groups extension is allowed to pickup by dialing '*8'.
Field Type: [0-9] [,-]

21.5.2.5.8 RX/TX



RX Wink:

Set timing parameters

Example:
- Pre-wink (50ms)
- Pre-flash (50ms)
- Wink (150ms)
- Receiver flashtime (250ms)
- Receiver wink (300ms)
- Debounce timing (600ms)

Field Type: Select box

RX Gain:

Receive signal decibel

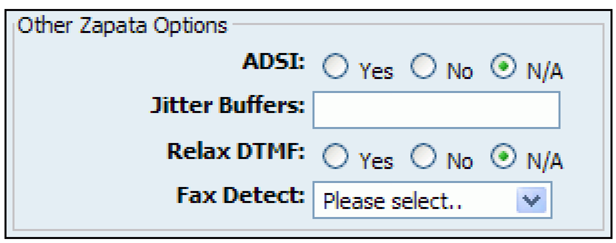
Example: 2
Field Type: [0-9]

TX Gain:

Transmit signal decibel

Example: 2
Field Type: [0-9]

21.5.2.5.9 Other Zapata Options



ADSI (Analog Display Services Interface):

Enable remotely controlling of screen phone with softkeys. (Only if you have ADSI compatible CPE equipment)

Example: Yes, No, N/A
Field Type: Option buttons

Jitter Buffers:

Configure jitter buffers. Each one is 20ms long

Example: 4
Field Type: [0-9]

Relax DTMF:

If you are having trouble with DTMF detection, you can relax the DTMF detection parameters

Example: Yes, No, N/A
Field Type: Option buttons

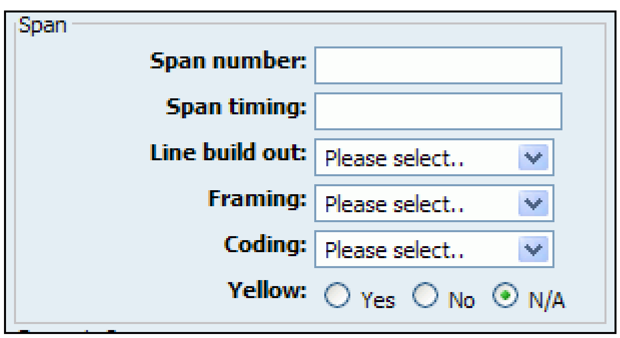
Fax Detect:

Enable fax detection

Example:
- both
- incoming
- outgoing
- no

Field Type: Select box

21.5.2.5.10 Span



Span number:

Number of the span

Example: 1
Field Type: [0-9]

Span timing:

How to synchronize the timing devices

Example:
- 0 - do not use this span as sync source
- 1 - use as primary sync source
- 2 - set as secondary and so forth

Field Type: [a-z]

Line build out:

Example:
- 0 db (CSU) / 0-133 feet (DSX-1)
- 133-266 feet (DSX-1)
- 266-399 feet (DSX-1)
- 399-533 feet (DSX-1)
- 533-655 feet (DSX-1)
- -7.5db (CSU)
- -15db (CSU)
- -22.5db (CSU)

Field Type: Select box

Framing:

How to communicate with the hardware at the other end of the line

Example:
- For T1: Framing is one of d4 or esf.
- For E1: Framing in one of cas or ccs.

Field Type: Select box

Coding:

How to encode the communication with the other end of line hardware.

Example:
- For T1: coding is one of ami or b8zs
- For E1: coding is one of ami or hdb3 (E1 may also need crc)

Field Type: Select box

Yellow:

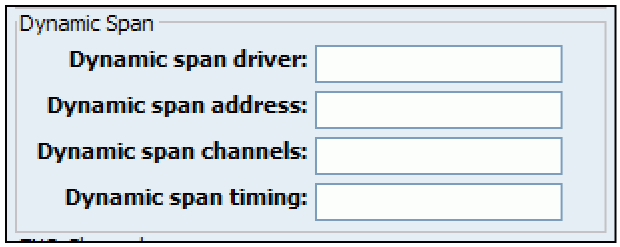
Whether yellow alarm is transmitted when no channels are open.

Example:

- Yes
- No
- N/A

Field Type: Option buttons

21.5.2.5.11 Dynamic Span



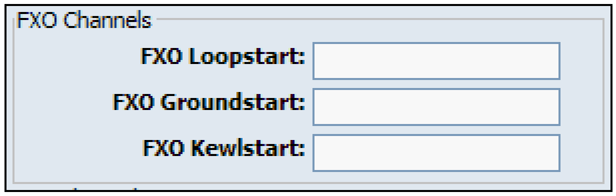
Dynamic span driver:
The name of the driver (e.g. eth)

Dynamic span address:
Driver specific address (like a MAC for eth).

Dynamic span channels:
Number of channels.

Dynamic span timing:
Sets timing priority, like for a normal span. Use "0" in order not to use this as a timing source, or prioritize them as primary, secondary, etc.

21.5.2.5.12 FXO Channels



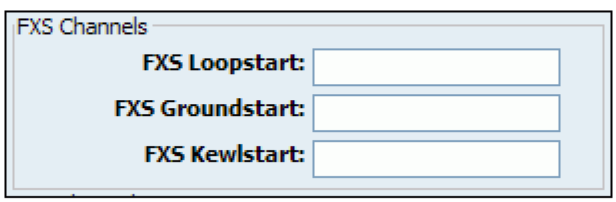
FXO Loopstart:
Channel(s) are signalled using FXO Loopstart protocol

FXO Groundstart:
Channel(s) are signalled using FXO Groundstart protocol

FXO Kewlstart:
Channel(s) are signalled using FXO Kewlstart protocol

- 1 - for one card
- 1-2 - for two cards
- 1-3 - for three cards etc or
- 2-3 (If your card has modules in this order FXS, FXO, FXO, FXS)

21.5.2.5.13 FXS Channels



FXS Loopstart:
Channel(s) are signalled using FXS Loopstart protocol

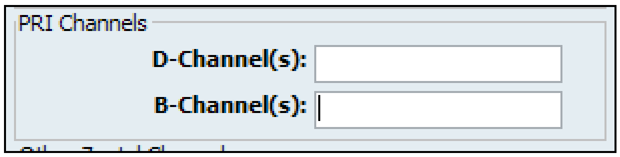
FXS Groundstart:
Channel(s) are signalled using FXS Groundstart protocol

FXS Kewlstart:
Channel(s) are signalled using FXS Kewlstart protocol

Values for above fields are set as follows:

- 1 - for one card
- 1-2 - for two cards
- 1-3 - for three cards etc or
- 1-4 (If your card has modules in this order FXS, FXO, FXO, FXS)

21.5.2.5.14 PRI Channels



D-Channel(s):

For example, every ISDN BRI card has 1 D- (control) channel

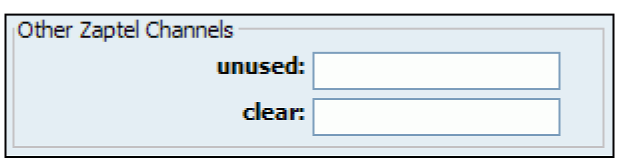
Example: 1
Field Type: [0-9]

B-Channels(s):

For example, every ISDN BRI card has 2 B- (data) channels

Example: 2
Field Type: [0-9]

21.5.2.5.15 Other Zaptel Channels



Unused:

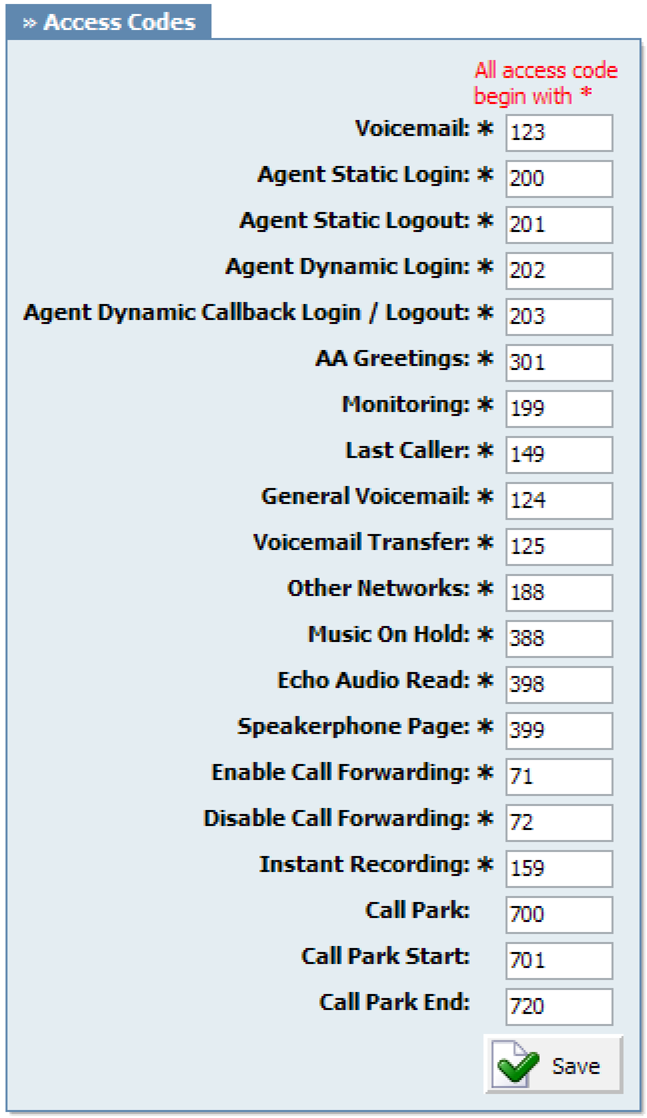
Example:
Field Type: [0-9]

Clear:

Example:
Field Type: [0-9]

21.6 Access Codes

Access codes provide system user with access to essential system or enhanced services



Voicemail:

Voice inbox access code. This number is dialed to access extension voice inbox (extension PIN required)

Example: From extension 1000 dial '*123' to access extension 1000 voice inbox. When asked for PIN, provide PIN set for this extension
Field Type: [0-9]

Agent Static Login:

Access code used for static agent login

Example: Dial '*200 + $QUEUE' from extension '1000' to login that extension into a specific Queue
Field Type: [0-9]

Agent Static Logout:

Access code used for static Agent logout

Example: Dial '*201 + $QUEUE' from extension '1000' to logout that extension from specific Queue
Field Type: [0-9]

Agent Dynamic Login:

Access code used for dynamic Agent login

Example: Dial '*202 + $AGENT_NUMBER' from extension '1000' to login into Queue as an Agent. Provide Agent PIN number when asked for it. Dynamic login means that connection has to be active all the time - Agent is not to hangup his connection or new login will be required.
Field Type: [0-9]

Agent Dynamic Callback Login/Logout:

Access code used for dynamic callback Agent login

Example: Dial '*203 + $AGENT_NUMBER' from extension '1000' to login into Queue as an Agent. Provide Agent PIN number when asked for it. Dynamic login means that connection has to be active all the time - Agent is not to hangup his connection or new login will be required.
Field Type: [0-9]

AA Greeting:

Access code for recording custom greetings used in IVR

Example: Dial '*301' from your extension and speak after the tone. When done, hangup the handset. Recorded file is saved as 'greeting-$DATA-***'. Navigate to 'System: Sound Files: G' and locate your sound file
Field Type: [0-9]

Monitoring:

Access code for monitoring active calls

Example: If Extensions '1000' and '1001' are in conversation, dial '*149 + 1000' to listen the ongoing conversation

Field Type: [0-9]

Last Caller:

Access code for dialling the last Extension that was calling you

Example: If Extension '1000' was the last extension calling you, dial '*149' from your Extension. Message 'The number to call your line was $EXTENSION. To call this number press 1' will be played. Press '1' dial this destination.
Field Type: [0-9]

General Voicemail:

Access code for general voice mailbox. This number is used for checking your voice inbox from any system Extension

Example: Dial '*124'. Enter your Extension number and PIN when asked for it
Field Type: [0-9]

Voicemail Transfer:

Access code for transferring active calls to any system voice box

Example: During active conversation dial '*125 + $EXTENSION' to transfer calling party to system $EXTENSION number voice box
Field Type: [0-9]

Other Networks:

Access code for accessing other PBXware networks

Example: Dial '*188 + $NETWORK + $EXTENSION' to dial number $EXTENSION or $NETWORK (e.g. '*188 8 1000'). NOTE: 'Trunk' and 'Other Network' must already be set
Field Type: [0-9]

Music On Hold:

Access code for playing Music On Hold sound files

Example: Dial '*388' to play 'default' Music On Hold class sound files
Field Type: [0-9]

Echo Audio Read:

Access code for echo audio test

Example: Dial '*398' and talk. Everything you say is returned back so the server response time can be checked
Field Type: [0-9]

Voicemail Transfer:

Access code for transferring active calls to any system voice box

Example: During active conversation dial '*125 + $EXTENSION' to transfer calling party to system $EXTENSION number voice box
Field Type: [0-9]

Speakerphone Page:

Access code for transmitting a message to multiple phones via their loudspeakers

Example: Dial '*399' to speakerphone all Extensions defined under Enhanced Services
Field Type: [0-9]

Enable Call Forwarding:

Access code for enabling Call Forwarding Enhanced Service

Example: Dial '*71 + $EXTENSION' to forward all calls to $EXTENSION number. This number can be local Extension or Proper/Mobile number (e.g. '*71 1001' or '*71 55510205')
Field Type: [0-9]

Disable Call Forwarding:

Access code for disabling Call Forwarding Enhanced Service

Example: dial '*72' to disable this service (Extension number not required)
Field Type: [0-9]

Instant Recording:

Access code for instant call recording

Example: During active conversation dial '*159' to start recording conversation
Field Type: [0-9]

Call Park:

Access code for parking active calls

Example: During active conversation dial '#700'. The call will be parked on first available Call Park Extension (e.g. 701). Parked call can be picked up from any system Extension by dialing parked Extension ('701')
Field Type: [0-9]

Call Park Start:

Start Extension for call parking service

Example: If set to '701' all calls will be parked on Extensions '701' to 'Call Park End'
Field Type: [0-9]

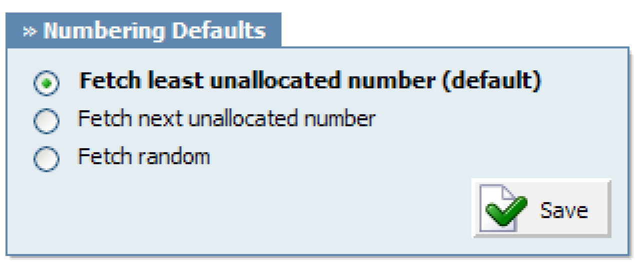
Call Park End:

End Extension for call parking service

Example: If set to '720' all calls will be parked on Extensions 'Call Park Start' to '720'
Field Type: [0-9]

21.7 Numbering Defaults

Numbering defaults sets the way PBXware will assign network numbers to Extensions, Conferences etc...



Fetch least unallocated number (default):
This option takes the least number on the system that is not used by the system and assign it to new Extension you're trying to create for example.

Fetch next unallocated number:

If the last allocated number was assigned to extension 2010, next IVR that you're trying to add for example will be given number 2011. System will not try to give you an unallocated number 1022 for example

Fetch random:
Use this option in case that you want to assign numbers in non sequential order. If you create new Queue for example, PBXware will assign it a number 2350 for example. And if you try to create new Extension right after that, PBXware might assign it a number 9838 for example.

21.8 Voicemail

Calls are diverted to Voicemail when user is unavailable, has the phone powered off or when a call is transferred to a voicemail by user. The phone alerts user to indicate the receipt of a message.

Once the user is transferred to party's voice box 'Please leave a detail message after the tone. If you would like to speak to the operator, press 0' message will be heard.

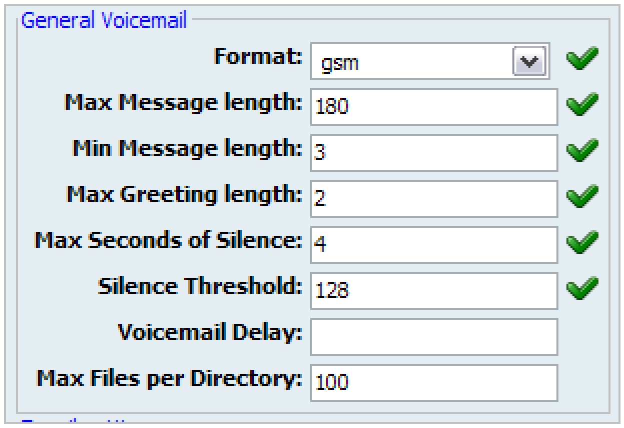
User has two options:
1. To leave a voice message that is ended by pressing # key or by hanging up, or
2. To reach an operator by dialing 0

If 0 is dialed 'Press 1 to accept this recording, otherwise please continue to hold' message will be heard.

User has two options:
1. Press 1 to save your message and dial the operator. 'Please hold while i try that extension' message played.
2. Continue to hold to delete your message and dial the operator. 'Message deleted, please hold while i try that extension' message played.

21.8.1 General Voicemail

General fields are most required by voicemail



Format:

Audio format voice messages are recorded in

Example: If 'wav49' is selected here, all voice messages will be saved in this format. See below for disk usage
Field Type: Select box

Max Message Length:

Maximum length of a voice message in seconds

Example: By default this field is set to '180' seconds (3 minutes)
Field Type: [0-9]

Min Message Length:

Minimum length of a voice message in seconds

Example: Default value set to '3' seconds. Messages that last less are discarded
Field Type: [0-9]

Max Greeting Length:

Maximum length in seconds of the user recorded voicemail greeting message

Example: Default values set to '60' seconds
Field Type: [0-9]

Max Seconds of Silence:

Maximum length of silence in a voice message in seconds

Example: Default value set to '10' seconds. Silence longer then set here will end a voice message
Field Type: [0-9]

Silence Threshold:

Silence detection threshold

Example: Default value set to '128'. Higher the number, more background noise is added
Field Type: [0-9]

Voicemail Delay:

Delay a number of seconds before asking user for 'Password'

Example: If you hear a partial sound file played asking user for password, set '1' or '2' here to add a second or two of silence before the sound file is played.
Field Type: [0-9]

Max Files per Directory:

Maximum number of voicemail messages per voicemail directory

Example: Each voice box has following directories (INBOX, Old, Work, Family, Friends, Cust1, Cust2, Cust3, Cust4, Cust5). Set this field to '100' to allow a 100 voice messages per each voice directory
Field Type: [0-9]

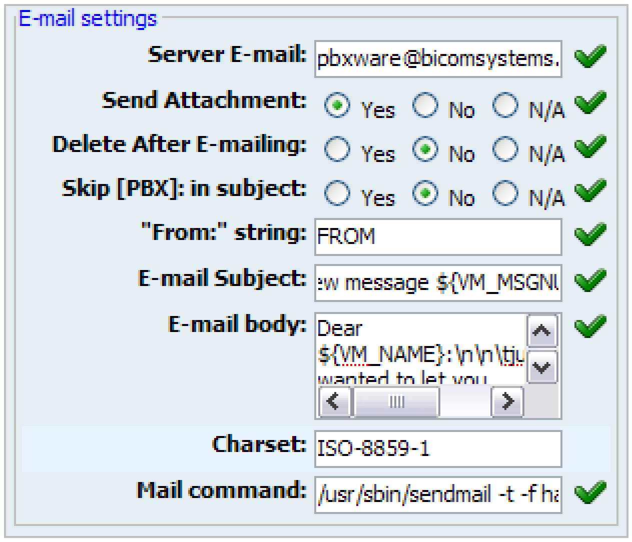
Disk Space Used By Voicemail Recording

With continuously tone 60 seconds:
- wav49 = 91.0kb
- wav = 863.0kb
- gsm = 91.0kb

With continuously silent tone (without sound) 60 sec:
- wav49 = 0.38kb
- wav = 3.0kb
- gsm. = 0.32k b

21.8.2 E-mail Settings

Customize the display of emails that notify user of new voicemail messages.



Server E-mail:

This email address is used to identify from whom the email came from

Example: If this field is set to 'pbxe@domain.com' in email header the following line is added '"$FROM": string <pbxe@domain.com>'
Field Type: [a-z][0-9]

Send Attachment:

Send the voice message as an attachment to user email.

Example: Once B gets the new voice message, if this option is set to 'Yes', the message sound file will be attached to the new voicemail notification email.
Field Type: Option buttons

Delete After E-mailing:

Delete voice message after sending it as an attachment to user email.

Example: Once B gets the new voice message, if this option is set to 'Yes', the message will be deleted from the voice box after it has been emailed to B.
Field Type: Option buttons

Skip[PBX]: in subject:

Should 'PBX' be skipped from voicemail title

Example: If set to 'No' - 'Subject: [PBX']: New message M in mailbox B' will be displayed in email subject line
Field Type: Option buttons

"From:" string:

Override a portion of the From: line in the voicemail notification email

Example: If this option is set to 'MARS', then email will have one additional line in its header 'From: MARS'
Field Type: [a-z][0-9]

E-mail Subject:

Customize voicemail notification email subject

Example: Use custom variables '${VM_MSGNUM}' for message number and '${VM_MAILBOX}' to create a custom email subject. 'PBXware: New message ${VM_MSGNUM} in mailbox ${VM_MAILBOX}' for example. Email subject would look like this 'PBXware: New message 3 in mailbox 1002'
Field Type: [a-z][0-9]

E-mail body:

Customize voicemail notification email body

Example: Use custom variables '${VM_NAME}' for User name, '${VM_MAILBOX}' for mailbox number, ${VM_DATE} for voicemail date and ${VM_DUR} for voicemail duration to create custom email body. For example

Dear, ${VM_NAME}\n\nYou have a new voicemail message on:\n\nmailbox: ${VM_MAILBOX}\n\nleft at: ${VM_DATE}\n\n${VM_DUR} long.
Field Type: [a-z][0-9]

Charset:

Character set notification email will be encoded with

Example: ISO-8859-1
Field Type: [a-z][0-9]

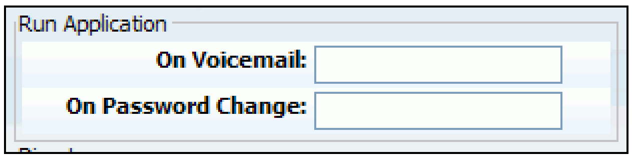
Mail command:

Overrides the default mailer command

Example: Default value '/usr/sbin/sendmail -t'
Field Type: [a-z][0-9]

21.8.3 Run Application

Run custom applications on certain voicemail actions



On Voicemail:

Run custom application when new voicemail is received

Example: Set this field to '/usr/bin/myapp' to execute 'myapp' application when new voicemail arrives
Field Type: [a-z][0-9]

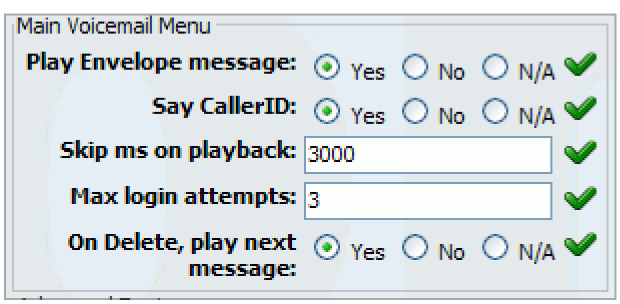
On Password Change:

Run custom application when voicemail password is changed

Example: Set this field to '/usr/bin/myapp' to execute 'myapp' application when voicemail password changes
Field Type: [a-z][0-9]

21.8.4 Main Voicemail Menu

Edit settings for main voicemail menu.



Play Envelope message:

Announces the Date/Time and the Extension number from which the message was recorded.

Example: Once voice box is checked for new messages, if this option is set to 'Yes', 'Received at {$DATE}. From phone number {$NUMBER}' will be played, giving more details about the message originator.
Field Type: Option buttons

Say Caller ID:

Announce the extension number from which the voice message has been recorded.

Example: If this option is set to 'Yes', when checking voicemail, 'From phone number {$NUMBER}' message will be heard.
Field Type: Option buttons

Skip ms on playback:

Interval in milliseconds to use when skipping forward or reverse while a voicemail message is being played

Example: If this field is set to '3000', when listening to voice message skip 3 seconds on rewind/fast forward
Field Type: [0-9]

Max login attempts:

Maximum number of login retries before user gets disconnected

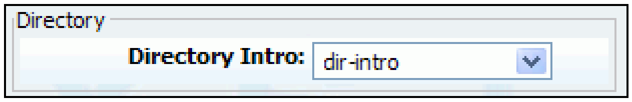
Example: By default this field is set to '3'. After 3 unsuccessful login attempts user gets disconnected
Field Type: [0-9]

On Delete, play next message:

After a voice message has been deleted, should the system automatically play the next message from voice inbox

Example: Select 'Yes' to automatically playback the next voice message after you've deleted the old one
Field Type: Option buttons

21.8.5 Directory

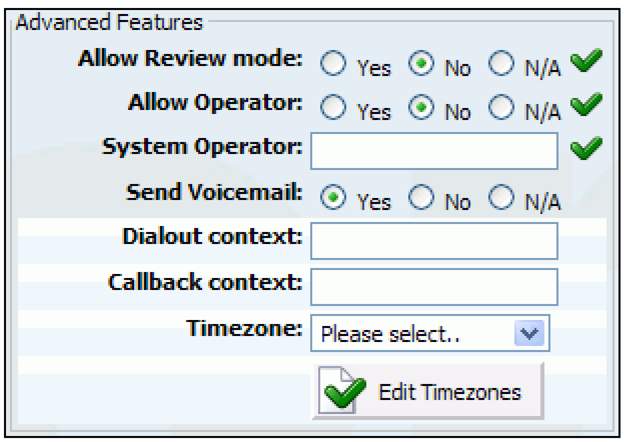


Directory Intro:

Override the directory intro file

Example: Default value 'dir-intro'
Field Type: Select box

21.8.6 Advanced Features



Allow Review mode:

Allow B to review the voice message before committing it permanently to A's voice box.

Example: B leaves a message on A's voice box, but instead of hanging up, he presses '#'. Three options are offered to B:

- Press 1 to accept this recording
- Press 2 to listen to it
- Press 3 to re-record your message

Field Type: Option buttons

Allow Operator:

Allow B to reach an operator from within the voice box.

Example: B leaves a message on A's voice box, but instead of hanging up, B presses '#'.
- 'Press 0 to reach an operator' message played (Once '0' is pressed, user is offered the following options):
  - Press 1 to accept this recording (If selected, 'Your message has been saved. Please hold while I try that extension' is played and operator is dialed)
  - Or continue to hold (If B holds for a moment, 'Message deleted. Please hold while I try that extension' is played and operator is dialed)

Field Type: Option buttons

System Operator:

Local extension number that acts as an operator.

Example: If A's voice box has an option 'Allow Operator' set to 'Yes', all users dialing '#0' inside the voice box will reach this operator extension.
Field Type: [0-9]

Send Voicemail:

Change context to send voicemail from

Example: Select 'Yes' to enable and provide new values to 'Dialout context' and 'Callback context' fields
Field Type: Option buttons

Dialout context:

Context to dial out from

Example: Set this field to 'fromvm' for example
Field Type: [a-z][0-9]

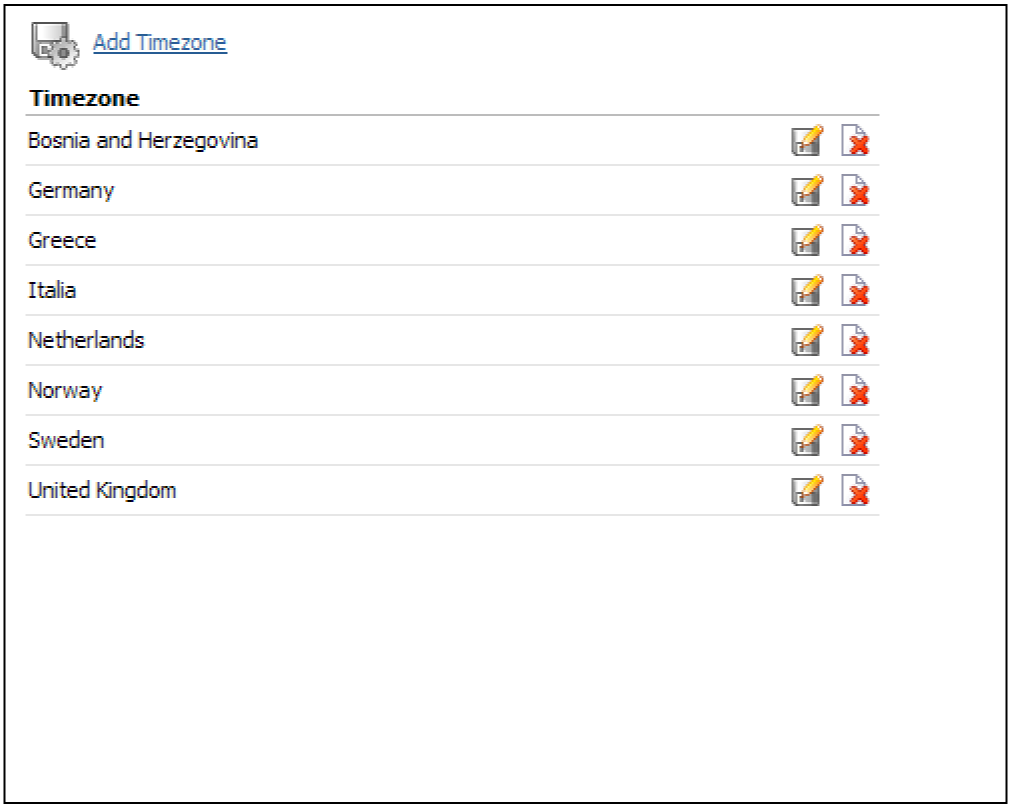
Callback context:

Context to call back

Example: Set this field to 'tomv'. If not listed, calling the sender back will not be permitted
Field Type: [a-z][0-9]

Add/Edit Timezones 469

21.8.6.1 Edit Timezones

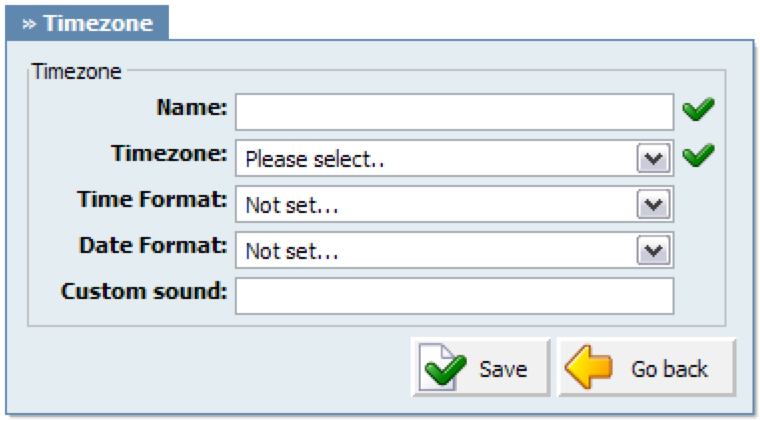


Timezone:
Timezone name
Example: Bosnia and Herzegovina
Field Type: Display

Edits the timezone configuration
Example: Click to edit timezone configuration
Field Type: Button

Deletes a timezone from the system
Example: Click to delete a timezone from the system
Field Type: Button

21.8.6.1.1 Add/Edit Timezone



Name:

Unique timezone name

Example: Name provided here will be visible when setting correct voicemail timezone. Type 'Zenica' here for example
Field Type: [a-z][0-9]

Timezone:

Set the correct timezone

Example: If you have set 'Name'='Zenica' (a town in Bosnia) select the closes timezone to Zenica here (e.g. 'Europe/Sarajevo')
Field Type: Select box

Time format:

Set the appropriate time format

Example: Depending on selected 'Timezone' you may choose between the following options:

- 12 Hour clock
- 12 Hour clock including minute
- 12 Hour clock AM/PM
- 12 Hour clock AM/PM, including minute
- 24 Hour clock
- 24 Hour clock including minute
- AM/PM 12 hour syntax
- Dutch syntax
- German syntax

- Greek syntax
- Italian syntax
- Norwegian syntax
- Swedish syntax

Field Type: Select box

Date format:

Set the correct date format

Example: Depending on selected 'Timezone' you may choose between the following options

- Month/Day/Year
- Day of Week/Month/Day/Year
- Day/Month/Year
- Day of Week/Day/Month/Year

Field Type: Select box

Custom sound:

This file is played before the voicemail arrival time

Example: Enter sound file name, without the extension (e.g. 'arlington') here
Field Type: [a-z][0-9]

21.8.7 Mailbox Options

Calls are diverted to Voicemail when user is unavailable, has the phone powered off or when a call is transferred to a voicemail by user. The phone alerts user to indicate the receipt of a message.

Once the user is transferred to party's voice box 'Please leave a detail message after the tone. If you would like to speak to the operator, press 0' message will be heard.

User has two options:
1. To leave a voice message that is ended by pressing # key or by hanging up, or
2. To reach an operator by dialing 0

If 0 is dialed 'Press 1 to accept this recording, otherwise please continue to hold' message will be heard.

User has two options:
1. Press 1 to save your message and dial the operator. 'Please hold while i try that extension' message played.
2. Continue to hold to delete your message and dial the operator. 'Message deleted, please hold while i try that extension' message played.

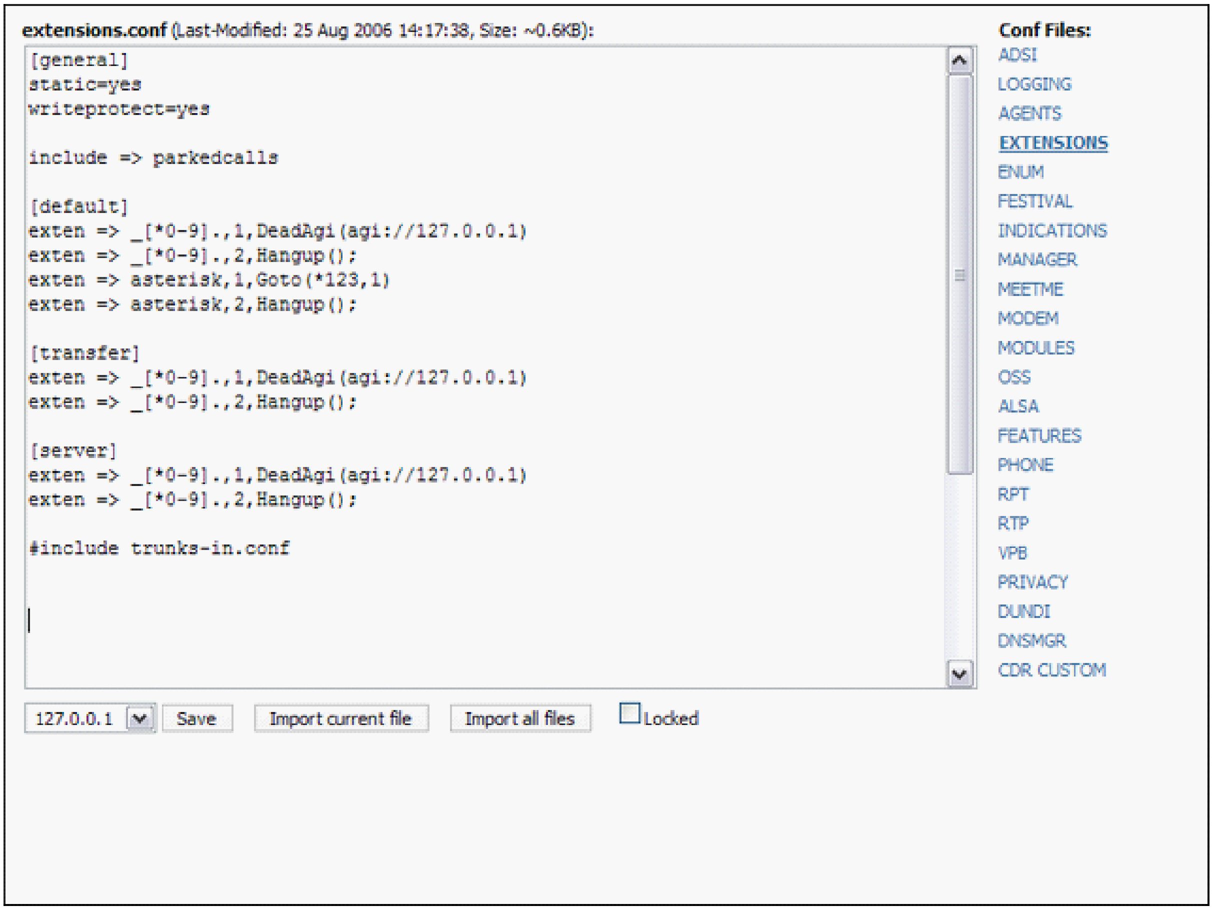
1 Read voicemail messages
2 Change folders
    0 Mailbox options
    1 Record your unavailable message
    2 Record your busy message
    3 Record your name
    4 Record your temporary message (new in Asterisk v1.2)
    5 Change your password
    ***** Return to the main menu
3 Advanced options (with option to reply; introduced in Asterisk CVS Head April 28, 2004 with 'enhanced voicemail')
    1 Reply
    2 Call back(1)
    3 Envelope
    4 Outgoing call(1)
4 Play previous message
5 Repeat current message
6 Play next message
7 Delete current message
8 Forward message to another mailbox
9 Save message in a folder
***** Help; during msg playback: Rewind
Exit; during msg playback: Skip forward

*** *** Help
*** #** Exit

After recording a message (incoming message, busy/unavailable greeting, or name)
1 Accept
2 Review
3 Re-record
0 Reach operator(1) (not available when recording greetings/name)

21.9 Configuration Files

System configuration files are accessible through this section. Only trained users should modify configuration files.



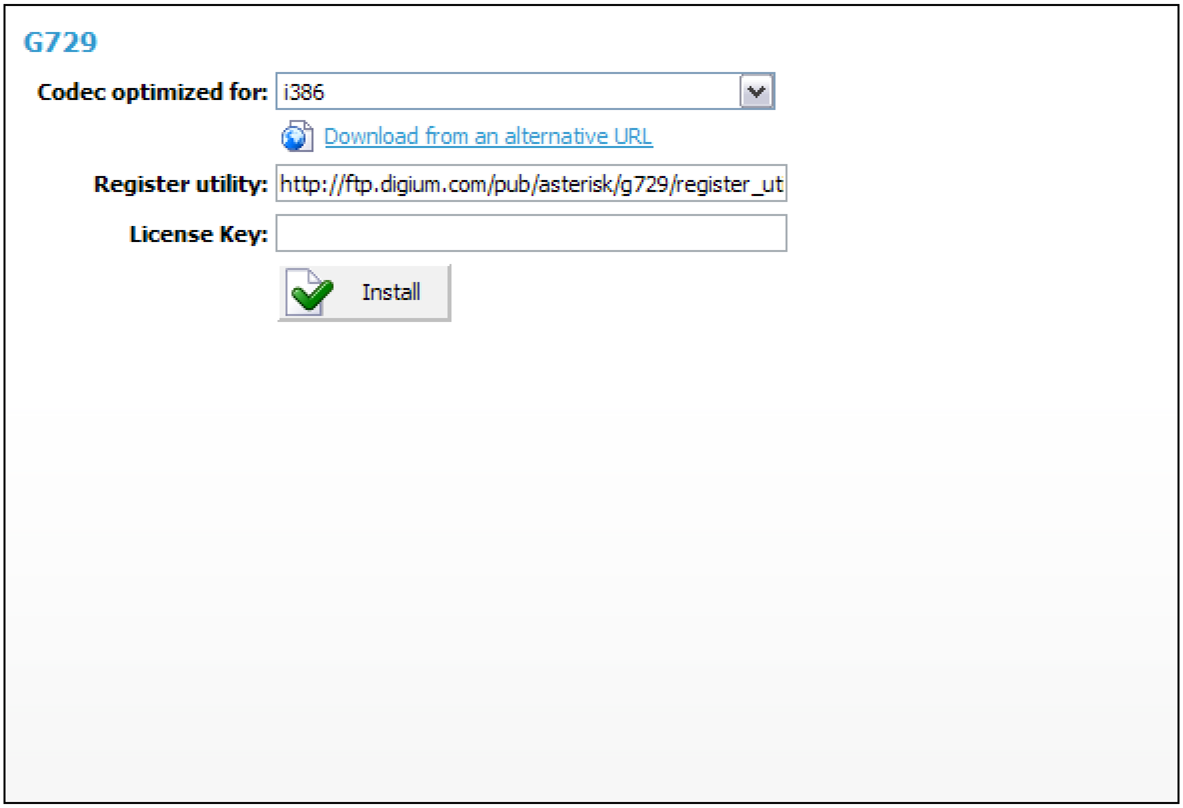
First select a configuration file from the right hand side navigation (e.g. FEATURES). The file will open in big text box from where it can be modified. Once the file is changed, click on 'Save' button.

If there is a message saying 'Configuration files not updated' visible, click on 'Import current file' or 'Import All Files' button to update the files. This error usually happens if configuration file is changed through system console (shell).

NOTE: Once any kind of file update is done be sure to Reload PBXware to apply the changes.

21.10 g729

G.729 codec can be easily downloaded and installed on PBXware by supplying just a few essential details.



Codec optimized for:

Select the server architecture.

Example: Select the type of server processor
Field Type: Select box

Register Utility:

Path to codec's register utility, required for codec install

Example: Please note that the default path leads to 32-bit software architecture. For 64-bit please visit http://ftp.digium.com/pub, navigate to desired version and paste the new destination into this field.
Field Type: [a-z] [0-9] [:/_.-]

http://ftp.digium.com/pub/asterisk/g729/register_utility/32-bit/register

License Key:

Codec license key.

Example: Provide the codec license key here.
Field Type: [a-z] [0-9] [_.-]

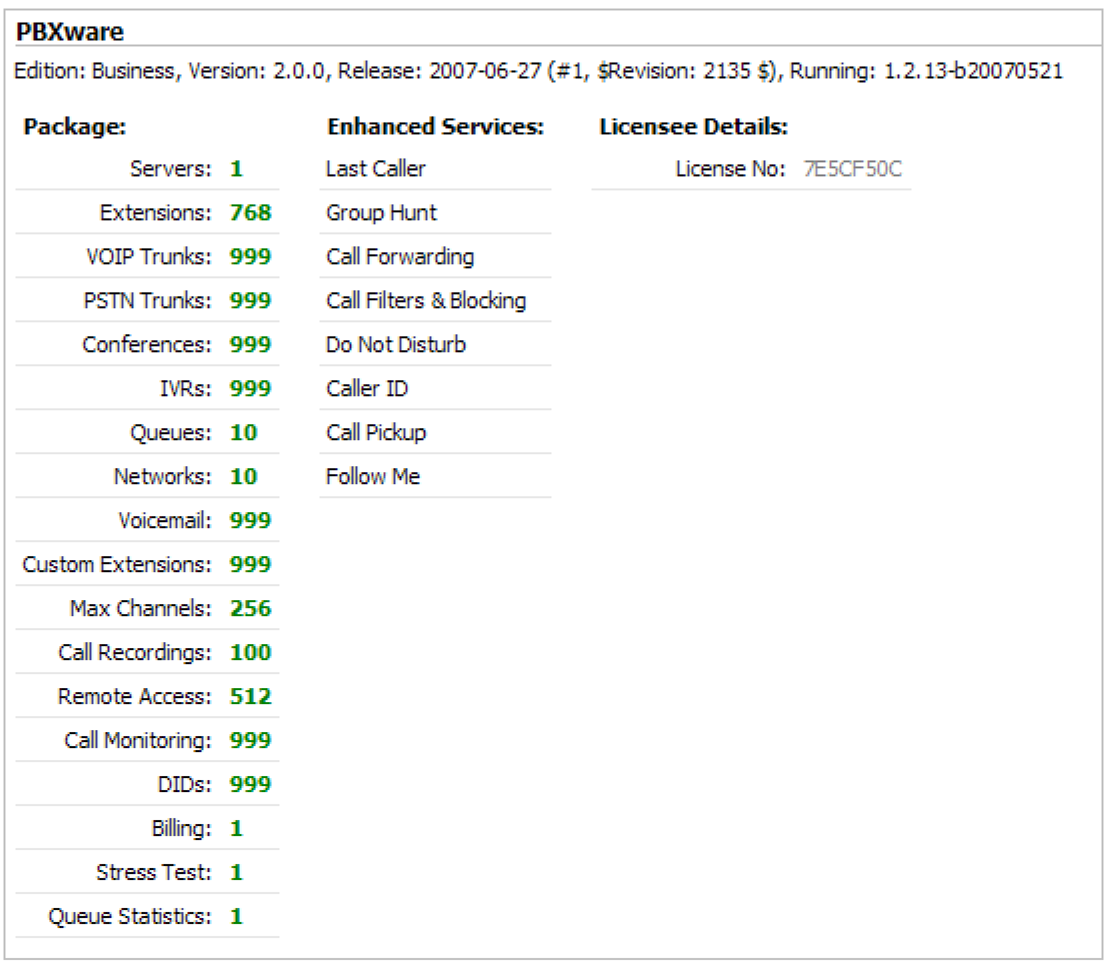
Install:

Install button

Example: After all necessary details are provided with the correct data(valid urls) click this button to install the g729 codec.
Field Type: Command button

21.11 About

About section displays systems edition, version, release date and licensing information.



PBXware:
This line identifies the PBXware version, release date (Revision) and Asterisk running
Example: Edition: Business, Version: 2.0.0, Release: 2007-06-27 (#1, $Revision: 2135 $), Running: 1.2.13-b20070521
Field Type: Display

Package:
Package information displays a number of system Extensions, Trunks, Conferences ...
Example: Servers: 1, Extensions 768...
Field Type: Display

Enhanced Services:
This line identifies system Enhanced Services
Example: Last Caller, Group Hunt, Call Forwarding...
Field Type: Display

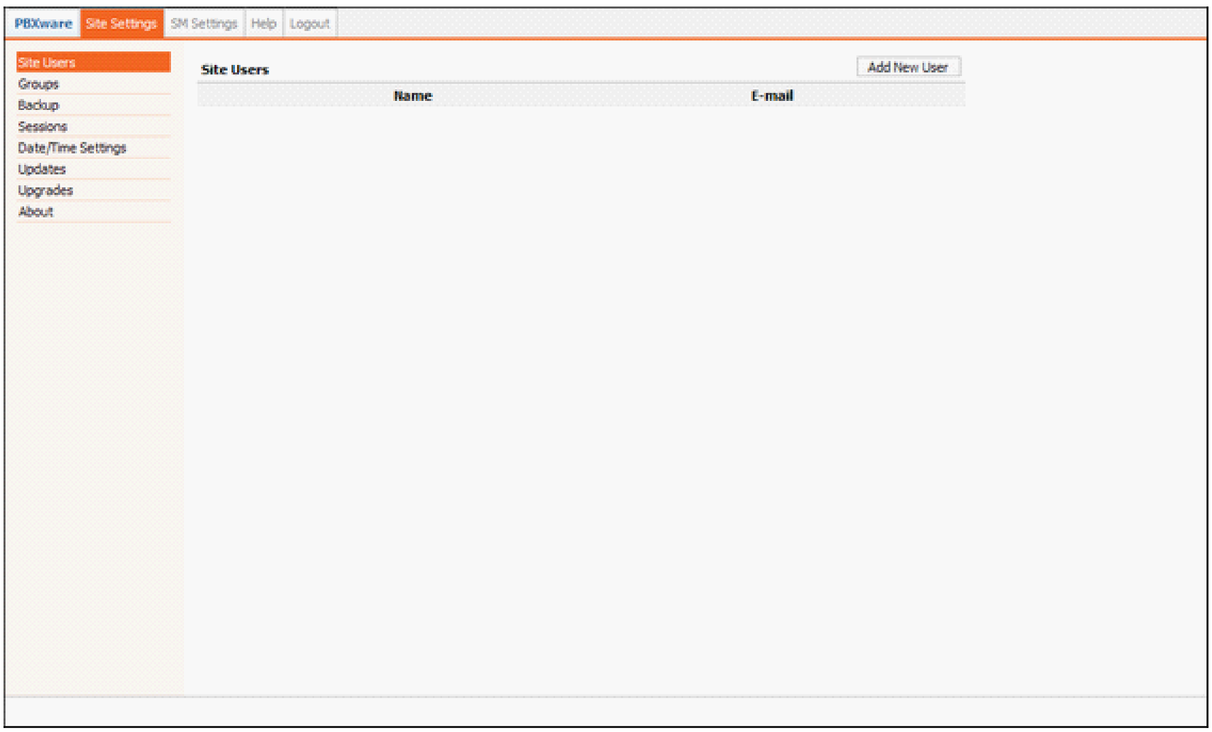
License Details:
This line displays license details. NOTE: If 'Branding' is enabled, only license number is visible
Example: License No: 7E5CF50C
Field Type: Display

Part XXI

22 Site Settings

Site settings set options such as site users, user groups, backup, updates and upgrades options

In this chapter we will cover:

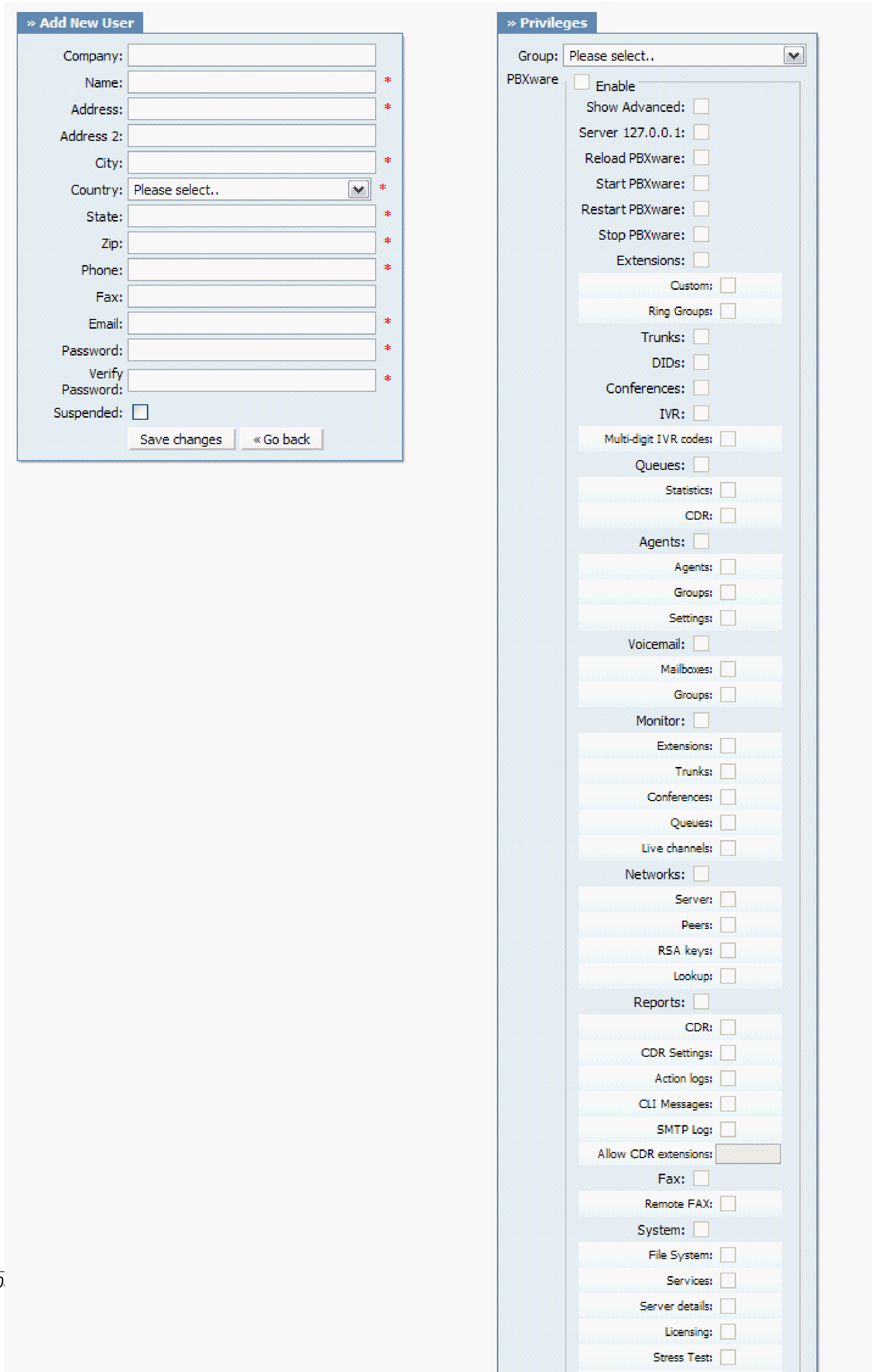


22.1 Site Users

Site users are allowed to login into the system interface in order to perform a specific function according to granted permissions. Each user belongs to a user group. Each group permissions are pre-set in order to allow unified access and permission control.

User can have access to any application or part of that application depending on permissions granted. It is highly recommended to add/edit groups before adding new users.

Add/Edit Users:

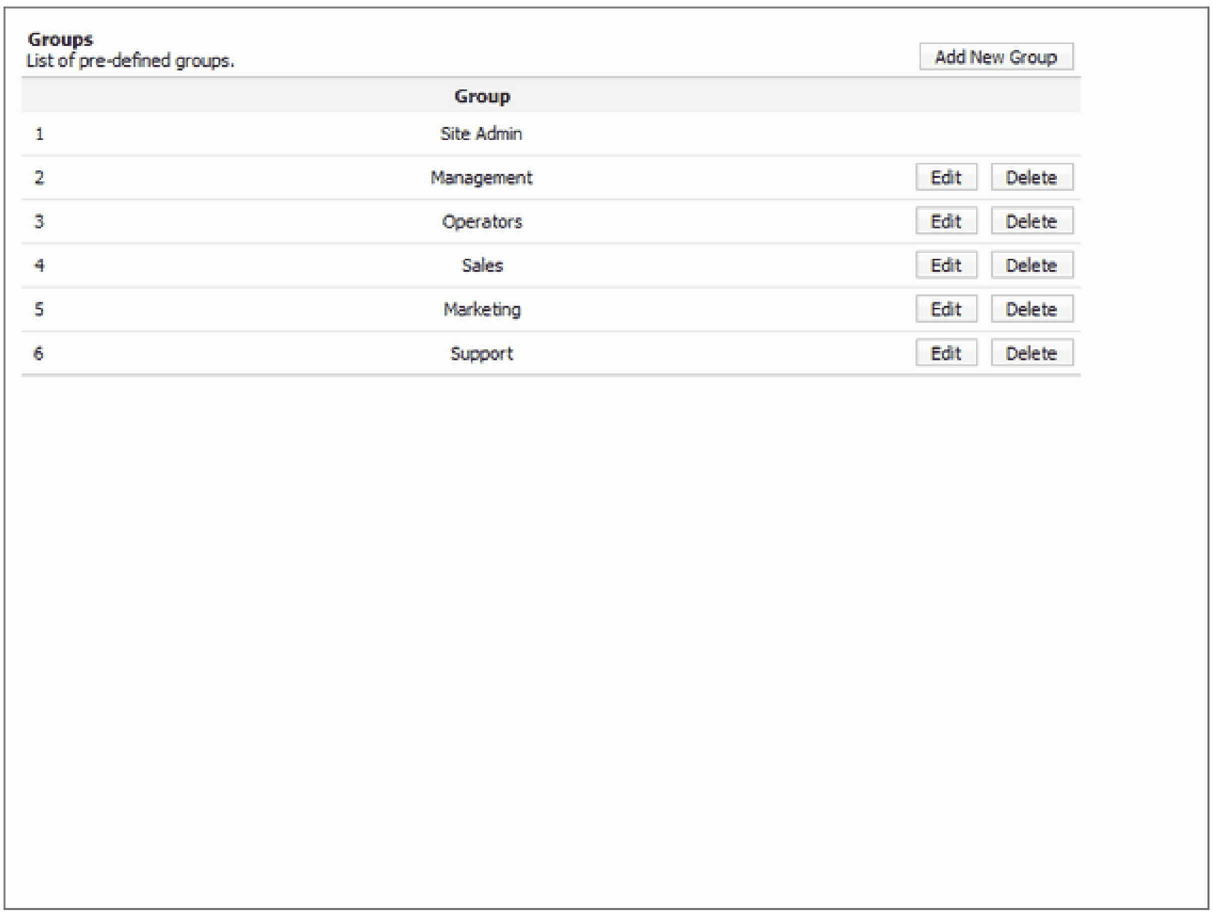

» Add New User
Company:
Name: *
Address: *
Address 2:
City: *
Country: Please select.. *
State: *
Zip: *
Phone: *
Fax:
Email: *
Password: *
Verify Password: *
Suspended:
Save changes
« Go back
» Privileges
Group: Please select..
PBXware
Enable
Show Advanced:
Server 127.0.0.1:
Reload PBXware:
Start PBXware:
Restart PBXware:
Stop PBXware:
Extensions:
Custom:
Ring Groups:
Trunks:
DIDs:
Conferences:
IVR:
Multi-digit IVR codes:
Queues:
Statistics:
CDR:
Agents:
Agents:
Groups:
Settings:
Voicemail:
Mailboxes:
Groups:
Monitor:
Extensions:
Trunks:
Conferences:
Queues:
Live channels:
Networks:
Server:
Peers:
RSA keys:
Lookup:
Reports:
CDR:
CDR Settings:
Action logs:
CLI Messages:
SMTP Log:
Allow CDR extensions:
Fax:
Remote FAX:
System:
File System:
Services:
Server details:
Licensing:
Stress Test:

Add/Edit User
User fields are standard fields required to be entered in order for system to allow user access to various applications. In addition users status can be changed to "suspended" by ticking check box and pressing "save"

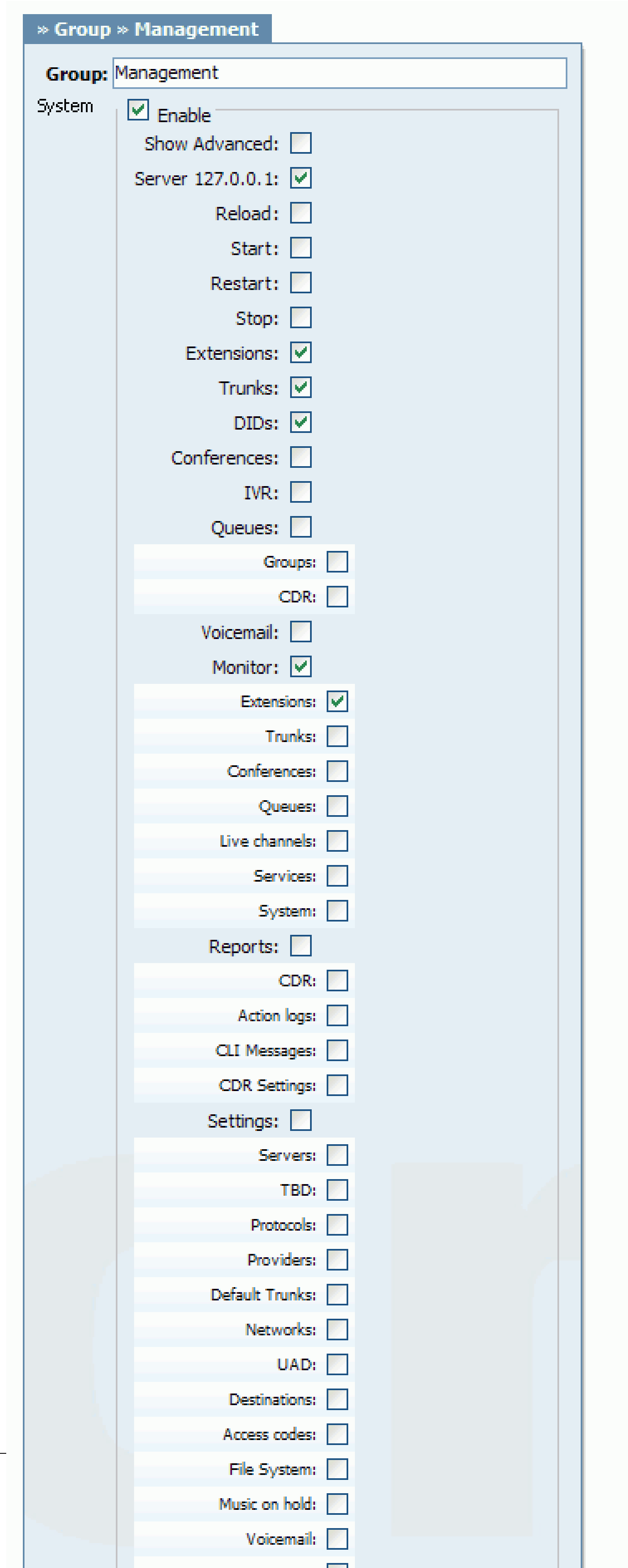
Privileges
This sections allows user to be assigned to a group for permissions purposes.

22.2 Groups

Groups allow for a unified permission system allowing users access to various applications or part of the applications. System is pre set with "common" groups: Sales, Support, Accounts and Billing, Management.



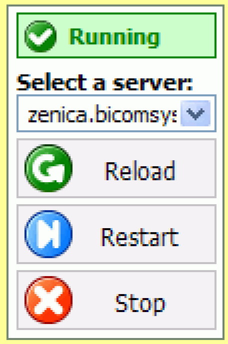
Each site can edit existing or ad new groups as per their requirements by clicking on appropriate action buttons. During add/edit permissions and group name is available for edit.


» Group » Management
Group: Management
System
Enable
Show Advanced:
Server 127.0.0.1:
Reload:
Start:
Restart:
Stop:
Extensions:
Trunks:
DIDs:
Conferences:
IVR:
Queues:
Groups:
CDR:
Voicemail:
Monitor:
Extensions:
Trunks:
Conferences:
Queues:
Live channels:
Services:
System:
Reports:
CDR:
Action logs:
CLI Messages:
CDR Settings:
Settings:
Servers:
TBD:
Protocols:
Providers:
Default Trunks:
Networks:
UAD:
Destinations:
Access codes:
File System:
Music on hold:
Voicemail:

Show Advanced
If this check box is ticked, all users belonging to this group will be able to view and edit all advanced options fields within the system.

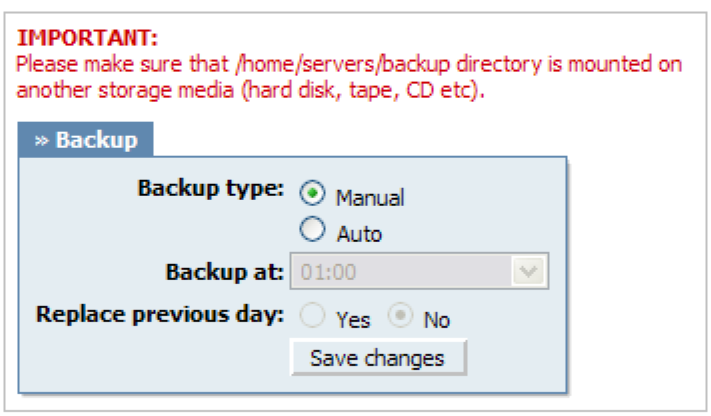
Server
If this check box is ticked, all user belonging to this group will have administration access to ticked server. This allows network administrator to delegate the administration across the organization.

Reload/Start/Restart/Stop PBXware button are options that control PBXware actions. If enabled on the right side of browser interface these controls will be displayed.



22.3 Backup

System backups copy all system data to /home/servers/backup on pre-set time of the day. The location of backup files (/home/servers/backup) must be mounted on another storage media (hard disk, tape, CD, network share etc).



Backup Type
Backups can be either automated or manual. Automated backups will be performed at pre-set time of the day. It is up the administrator to perform manual backups.

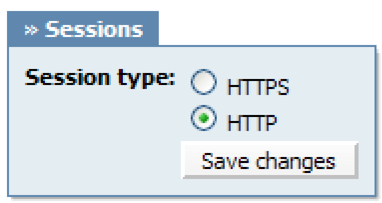
Backup at
Time at which backup will be performed

Replace previous day
If this vales is set to "Yes" the previous days backup will be deleted in order to save space on backup storage media. However, this does mean that at any time there will be only one days backup present. If this value set to "No", backups will not will be deleted but in this case storage media needs to have enough space available and it does need to be monitored regularly for available space.

WARNING:
If space is not available the backups will not be performed...

22.4 Sessions

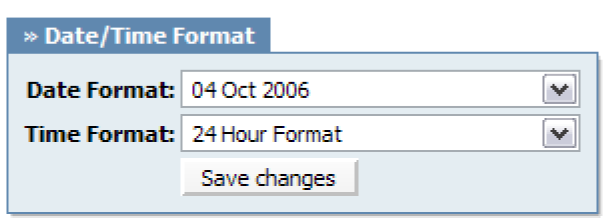
After an user is are logged into the system all data send between user and the system can be send in plain text or encrypted by industry standard SSL (secure layer socket). It is worth mentioning that using SSL uses much more resources hence the browser responses are slower.



Session Type
Available options:

- HTTPS - Encrypted SSL
- HTTP - Plain Text (Default)

22.5 Date/Time Settings



Date Format:

Set the proper date format used by the PBXware

Example: 04 Oct 2006
Field Type: Select box

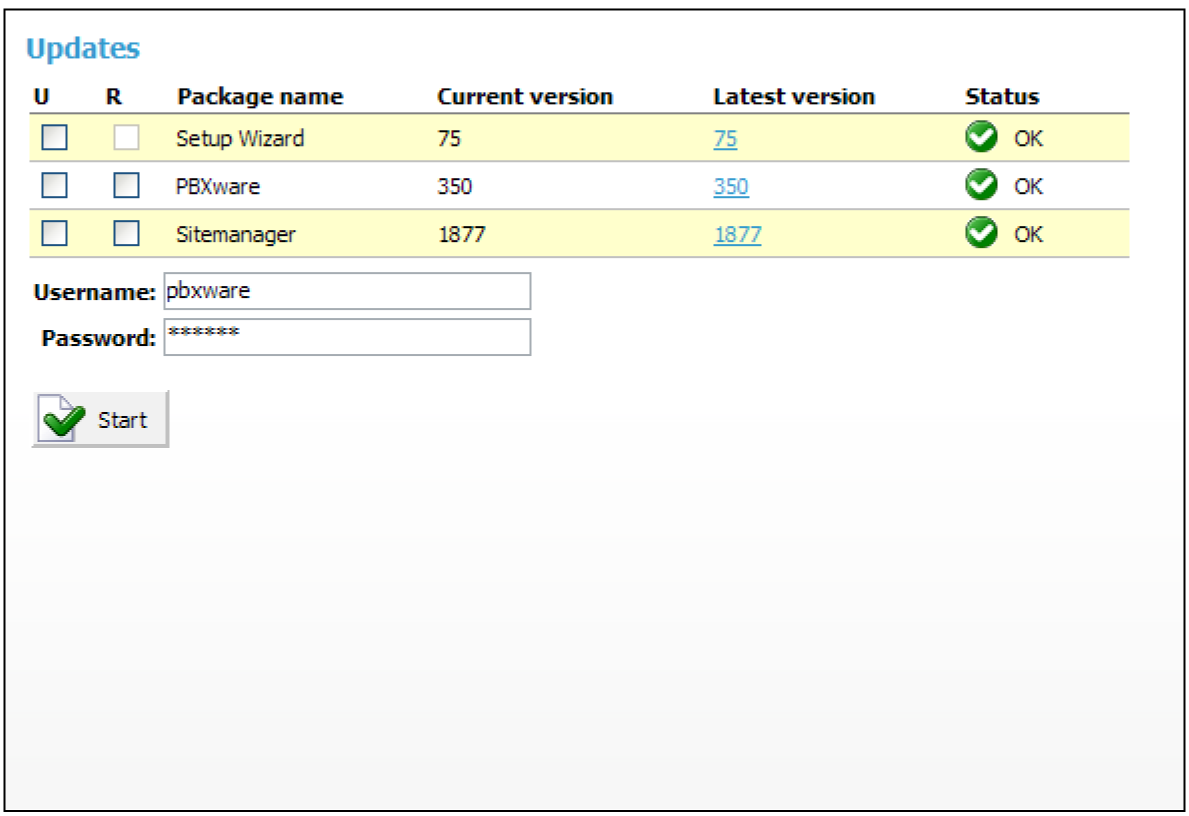
Time Format:

Set the proper time format used by the PBXware

Example: Select between 12/24 hour format
Field Type: Select box

22.6 Updates

This section allows an shortcut to licensing screen. It is useful if an license upgrade needs to be performed.



Update type:
Available options:
- Manual
- Auto (Has to be enabled for other fields to be enabled)

Username:
Username used for update.

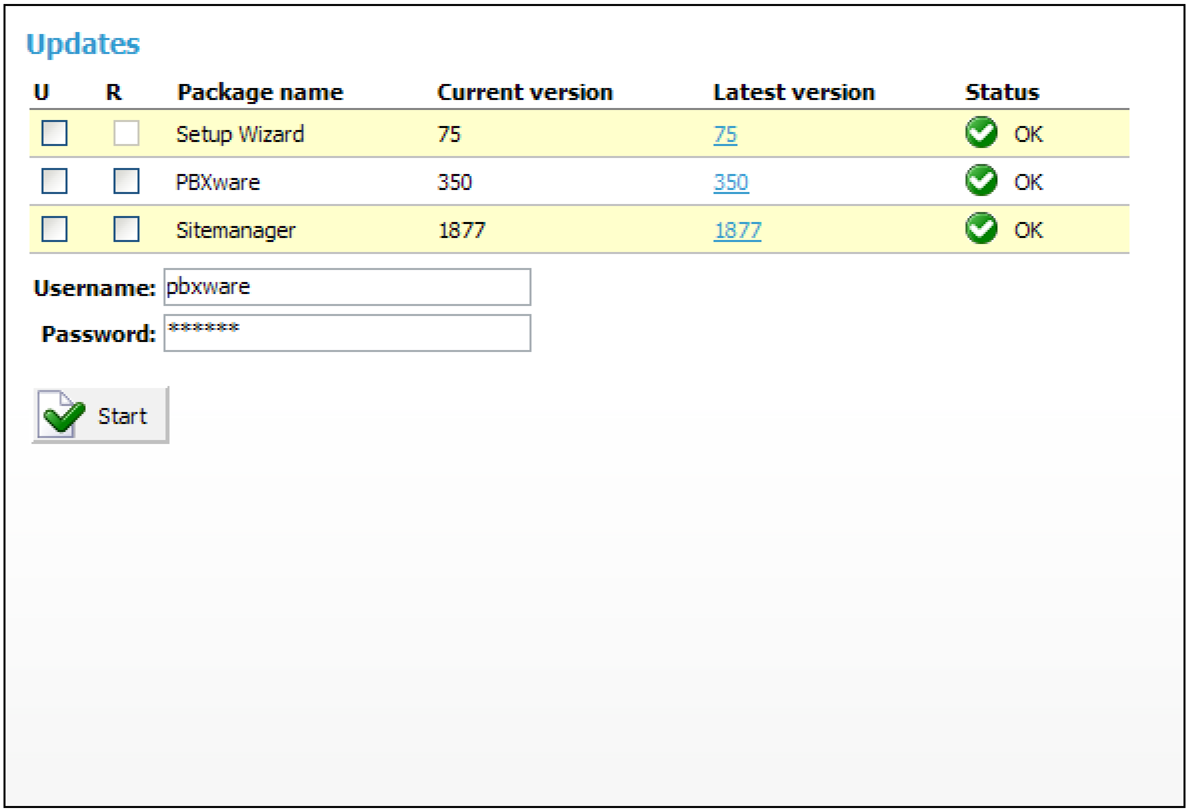
Password:
Password used by user for update.

Auto update type:
Select whether update should be done at certain time of the day or at regular time interval.
Available options:
- Time of the day
- Regular interval

Auto update interval:
Select time when update is done.

22.7 Upgrades

System is able to update it self at pre-set times of a day using included auto update daemon or alternatively a manual update can be performed by an authorized user.



Root Password / Confirm Root Password
System root password.

IMPORTANT:
This password is valid password for system (root), setup wizard and licensing upgrades entries.

Time zone
Time zone at which system is located or operating from.

Hostname
System Hostname

DHCP (Yes/No)
This is default value allowing the system to obtain its networking details from available DHCP server. However, setting DHCP to "No" allows for static network configuration.

22.8 About

About shows Site Manager current release date present on the system.

Site Manager

Release: 2006-08-29 (#1) $Revision: 1838 $

Part XXII

23 Troubleshooting

Link

Part XXIV

24 Howtos

Link

Part XXV

25 End Points

Link

Part XXV

26 Glossary

Glossary of Terms

Asterisk - Implementation of PBX on whose backbone our system is built

AGI - [Asterisk Gateway Plan] - Interface through which external programs control the dialplan

BRI [Basic Rate Interface] - ISDN configuration made out of 2 voice/data channels and 1 signalling channel

CLI [Command Line Interface] - Interface for interacting with the system using a command line

CSV [Comma Separated Value] - File format where columns are separated by comma ',' and rows by new line

DHCP [Dynamic Hosts Configuration Protocol] - Protocol for assigning different IP address to a device any time it connects to a network

DID [Direct Inward Dial] - Inbound line for dialing system destinations directly without the need for an operator

DNS [Domain Name Service] - Service which translates Internet domain names into IP addresses

DTMF [Dual Tone Multi Frequency] - A specific frequency consisting of two separate tones sent by UAD/ Phone each time a
key is pressed

DynDNS [Dynamic Domain Name Service] - Service used by clients on dynamic IP addresses which allows them to be contacted regardless of their current IP address

E.164 - Up to 15 digits international public telecommunication numbering plan. It consists of $COUNTRY_CODE + $NATIONAL_CODE + $SUBSCRIBER_NUMBER

IAX [Inter-Asterisk eXchange protocol] - Protocol for server-server or server-client connection

ISDN [Integrated Services Digital Network] - Digital voice and data transmission system over telephone wires

IVR [Interactive Voice Response] - System that manages incoming calls by playing available options and catching user response (pressed digit)

LAN [Local Area Network] - Communications network limited to immediate area

LCR [Least Cost Routing] - Diverts the calls to destination via the cheapest provider

WAN [Wide Area Network] - Geographically dispersed telecommunications network

MAC [Media Access Control] - Unique code (fingerprint) assigned with network devices

NAT [Network Address Translation] - IP packets rewriting technique used by multiple hosts on a private network to access Internet using a single IP address

PBX [Private Branch Exchange] - A smaller version of telephone company switch

PIN [Personal Identification Number] - Four digit security code required for accessing restricted system parts such as Voicemail, Enhanced Services, Conferences etc...

POSTFIX - Mail Transfer Agent used for routing/sending of system emails

PSTN [Public Switched Telephone Network] - The traditional, plain old telephony system

PRI [Primary Rate Interface] - ISDN configuration made out of 23/30 voice data channels and 1 signalling channel

RTP [Real-time transport protocol] - Internet standard for transporting real-time data (audio and video)

SIP [Session Initiated Protocol/Session Initiation Protocol] - Signaling protocol for Internet telephony

SMTP [Simple Mail Transfer Protocol] - Protocol used to send and receive email

UAD [User Agent Device] -Telephone

TTL [Time to Live] - Time in milliseconds system will wait for the response

Zaptel - Computer telephony hardware driver