XR1000
USER MANUAL

XORCOM

Risk-free Asterisk™

Document Information

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Contact Information

Xorcom Ltd., a contributing member of the Asterisk community, develops software and hardware for the open source Asterisk PBX environment. Our goal is to make Asterisk a friendly, easy–to–install and easy–to–use system, enabling fast and simple installation and configuration of PBXs of all sizes.

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support@xorcom.com

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INTRODUCTION

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Package Contents

Please ensure that the package contains the following:

- The XR1000 (10 inch or 19 inch width unit)
- External desk-top 12V 5A power supply unit
- Power cord
- Support hardware for 19" cabinet (for 19" full-size units only)
- The Xorcom CD-ROM
- The Getting Started leaflet

System Overview

The XR1000 is an embedded DSP-based Asterisk server. The device may be equipped with various combinations of up to 32 FXO and FXS ports, and with auxiliary ports for external appliances such as door locks and alarm systems. The XR1000 can support up to 32 concurrent calls and, in addition, can support SIP and IAX2 phones and trunks.

The XR1000 comes equipped with a reset-to-default-settings option that returns the appliance to the original factory configuration (see “Rescue, Restore, and Reflashing Firmware” on page 74).

In the event that the device has been damaged by improper modification—for example, by a programmer that make an erroneous code change even at the systems code level—the rescue option can save the XR1000 and return it to factory specifications.

For a full list of available XR1000 models, visit our web-site:

http://www.xorcom.com/products/xr1000/xr1000_models

Installation

To install the XR1000:

1. Select an appropriate location. The location must allow ample air circulation around both the power supply unit and the XR1000 box.

WARNING! *Never cover the power supply unit or the XR1000 box.*

WARNING! *Never use a power supply unit that was not supplied with the XR1000 box. Connecting other power supply units may damage the unit permanently and voids the warranty.*

2. For safety reasons, if the line wiring exits the building, it is essential to use the ground the unit.
   a. Connect a grounding tab to the grounding screw on the rear panel.
   b. Using 12-16 gauge wire, connect the grounding tab to a reliable ground.

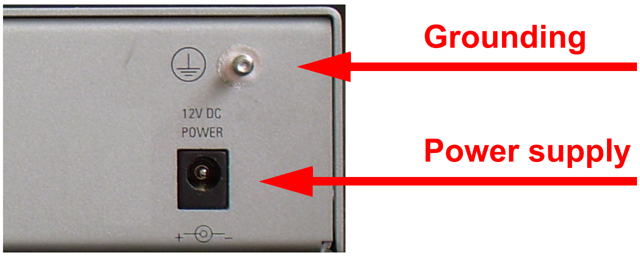


Figure 1: Grounding Screw on Rear Panel

3. Connect the power supply unit to the power supply input on the back of the unit.

WARNING! *Do not connect the device's LAN port to your office network. If your network is equipped with a DHCP server, this might cause network conflicts.*

4. Connect the device's WAN port, on the rear panel, to the office network or to your local router.

WARNING! *Regarding step #5, use a grounded power source.*

5. Connect the power supply unit to an electric outlet.
6. Wait about two minutes. The startup time will depend upon the number of telephony ports you have installed in your device.

   The FXS ports of the XR1000 will be operative about 30 seconds after the SYNC LED, located on the front panel, starts to blink.

7. Determine the IP address of your XR1000. If a DHCP server is available on your office network, an IP address has already been assigned to your XR1000. If not, configure the IP address manually (see “WAN Settings” on page 11).

   There are two ways to determine the IP address:

   Using an analog telephone
   Connect a phone to an FXS port and dial *66. The unit will announce the IP address.

   Using the Web interface
   To check the IP address via a browser:

   a. Using a standard Ethernet cable, connect a computer to the XR1000’s LAN port.

   b. Configure your computer to automatically obtain an IP address from a DHCP server. The XR1000 will assign the address 192.168.1.xxx to your computer.

   c. Start your Firefox Internet browser and enter 192.168.1.1 in the address field. The XR1000 web interface is displayed.

   d. Log-in to the unit. The default user name is **admin** and the default password is **123456**.

   e. On the **System Configuration** navigation pane, click **WAN** and configure the **WAN IP** settings. See “WAN Settings” on page 11 for a complete explanation of this procedure.

Note: Step 8 is possible only if your device is equipped with FXS ports.

8. Check the PBX functionality by completing one or more of the following tests.

   - Dial *65. The unit will announce the extension number. All FXS extensions are factory pre-configured. The default numbers begin at 401.
   - Make a call from one extension to another.
   - Dial *66. The unit will announce the WAN interface IP address.

9. Connect analog phones or lines to the XR1000.

   Connecting Analog Phones
   The XR1000 module supports:

   - Any standard analog phone
   - DTMF and pulse dialing
   - Caller ID
   - Message Waiting Indicator (MWI)
     A signal that activates the message light indicator—available on some telephone sets—when a message is available

   Full-size XR1000 units—with 24 or 32 FXS ports—have a Telco connection option. This option includes a special connector on the rear panel. The Telco connection enables the use of a single cable to connect all the FXS ports (up to 32) to the patch panel without having to assign an RJ-11 cable for each FXS port. At the same time, all the front panel LED indicators for each channel are available, and the RJ-11 connectors on the front panel enable easy access for testing and maintenance of each port.

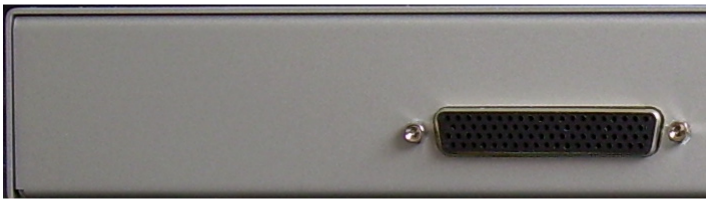


Figure 2: Telco Connection (Rear Panel)

Connecting Analog Lines

The XR1000 module supports standard PSTN lines.

⚠ ***WARNING!*** *When connecting the XR1000 to PSTN lines, ensure that the lines are protected. Use standard lightning protection devices such as gas discharge units or similar solid state, high-energy protection devices. In addition, ensure that the unit has been grounded using the grounding screw (see step #2 on page 3).*

Analog lines need to be tuned to adjust the port impedance to the telephone line impedance. Adjusting impedances can substantially reduce the echo that is generated by the interface of a digital 4-wire system to an analog 2-wire system. See “Tune FXO” on page 28.

LED Indicators

The LED sets on the front panel of a XR1000 include:

- 4 red LEDs (upper left) that indicate device status
- A green LED for each analog port
- 4 additional red LEDs that are not used in the current version

Device Status LEDs

Figure # 3 shows the status LEDs on analog XR1000 units.

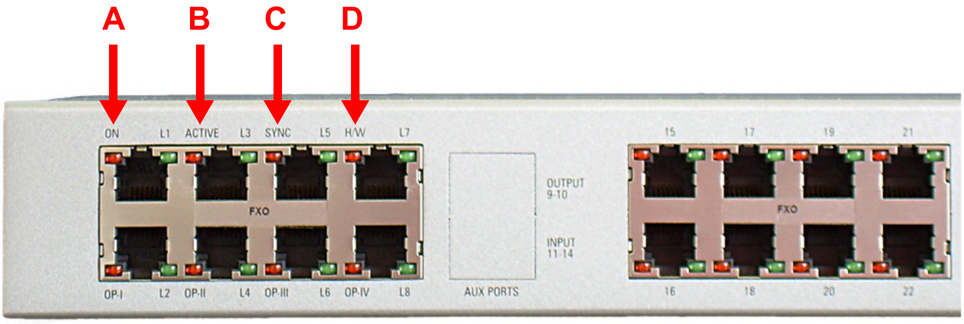


Figure 3: Analog Unit Device Status LEDs (red)

Table 1: Analog Unit Device Status LEDs

A	ON	Indicates that the unit is powered by the external desktop power supply
B	ACTIVE	Indicates that at least one XR1000 port (either FXS or FXO) is active
C	SYNC	Blinks when the XR1000 is up and has initialized the telephony ports.
D	H/W	Indicates hardware failure

Analog Line Status LEDs

The green LED on each RJ-11 port indicates analog line status:

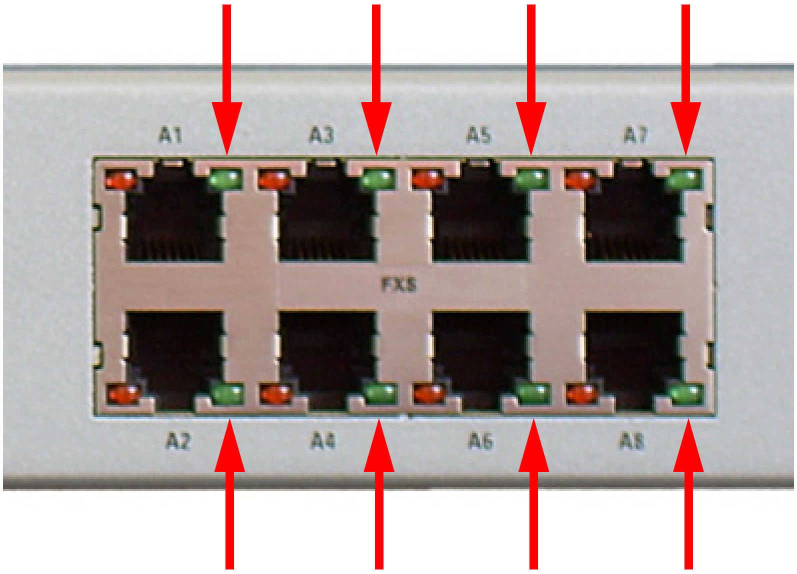


Figure 4: LEDs on RJ-11 Analog Ports

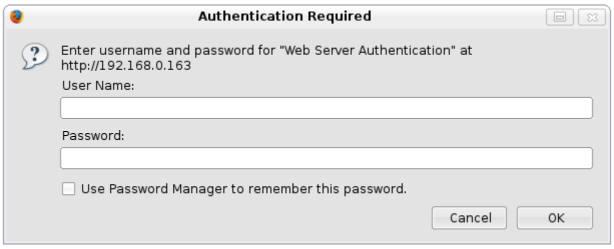
Table 2: Green Analog Line Status LEDs

Off	The port is not active.
On	The port is active, that is, off hook.
Blinking	Ringing, that is, either ring generation for FXS ports or ring detection for FXO ports

Log-in and Overview of the XR1000 Configurator

After installation is complete, log-on and configure the XR1000.

1. Start your Firefox Internet browser and enter the IP address of the XR1000 in the address field. The XR1000 log-on Authentication dialog is displayed.



Authentication Required

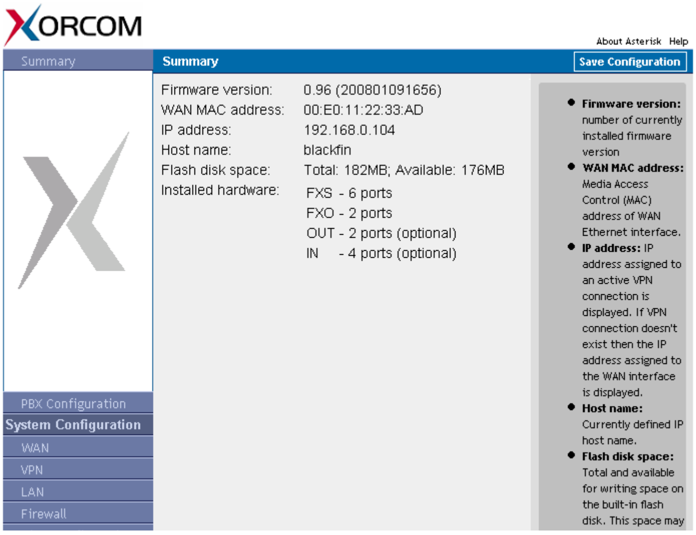
Enter username and password for "Web Server Authentication" at http://192.168.0.163

User Name:

Password:

□ Use Password Manager to remember this password.

2. Log-in to the unit. The default user name is **admin** and the default password is **123456**. The Summary screen is displayed.



Firmware version: 0.96 (200801091656)
WAN MAC address: 00:E0:11:22:33:AD
IP address: 192.168.0.104
Host name: blackfin
Flash disk space: Total: 182MB; Available: 176MB
Installed hardware: FXS - 6 ports
FXO - 2 ports
OUT - 2 ports (optional)
IN - 4 ports (optional)

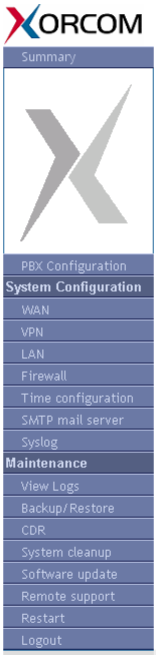
- **Firmware version:** number of currently installed firmware version
- **WAN MAC address:** Media Access Control (MAC) address of WAN Ethernet interface.
- **IP address:** IP address assigned to an active VPN connection is displayed. If VPN connection doesn't exist then the IP address assigned to the WAN interface is displayed.
- **Host name:** Currently defined IP host name.
- **Flash disk space:** Total and available for writing space on the built-in flash disk. This space may

The screen is divided into 4 sections:

Buttons and links in the blue title bar
The blue title bar, at the top of the screen, contains buttons and links, for example, the Save Configuration button and the Show Advanced Options link (see “Options” on page 70).

*WARNING! All changes must be saved on the Flash Memory prior to rebooting. Any changes that are not saved will be lost in the reboot process. This is done with the **Save Configuration** button located in the upper-right corner of the screen.*

The navigation panel on the left
The navigation panel (pictured on right) is divided into 3 sections, PBX configuration, System configuration, and Maintenance.



The configuration pane in the center
Changes to configuration are entered in the center pane.

Explanations of the configuration pane’s content on the right
These explanation are often enough to help you complete the procedure. For more detailed explanations, open the Help.

The PBX Configuration Panel opens in a separate window or tab of your Internet browser.

3. On the navigation pane, click **PBX Configuration**.

4. Configure the PBX. PBX configuration is fully documented beginning on page 33.

SECTION 2:

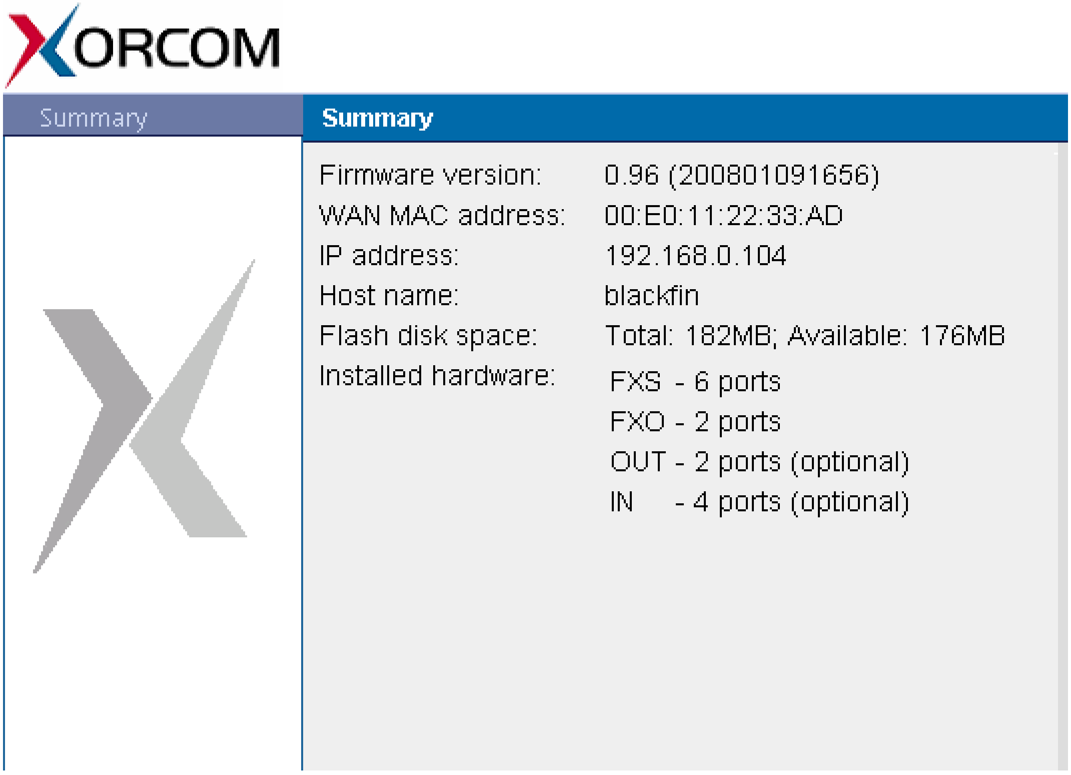
SYSTEM CONFIGURATION

The System Configuration menu includes the following:

System Summary

The System Summary provides an overview of the XR1000's configuration.

1. Log-in to the XR1000 Configurator (see "Log-in and Overview of the XR1000 Configurator" on page 7).
2. On the navigation pane, click **Summary**.



Summary

```
Firmware version:     0.96 (200801091656)
WAN MAC address:      00:E0:11:22:33:AD
IP address:           192.168.0.104
Host name:            blackfin
Flash disk space:     Total: 182MB; Available: 176MB
Installed hardware:   FXS  - 6 ports
                      FXO  - 2 ports
                      OUT  - 2 ports (optional)
                      IN   - 4 ports (optional)
```

WAN Settings

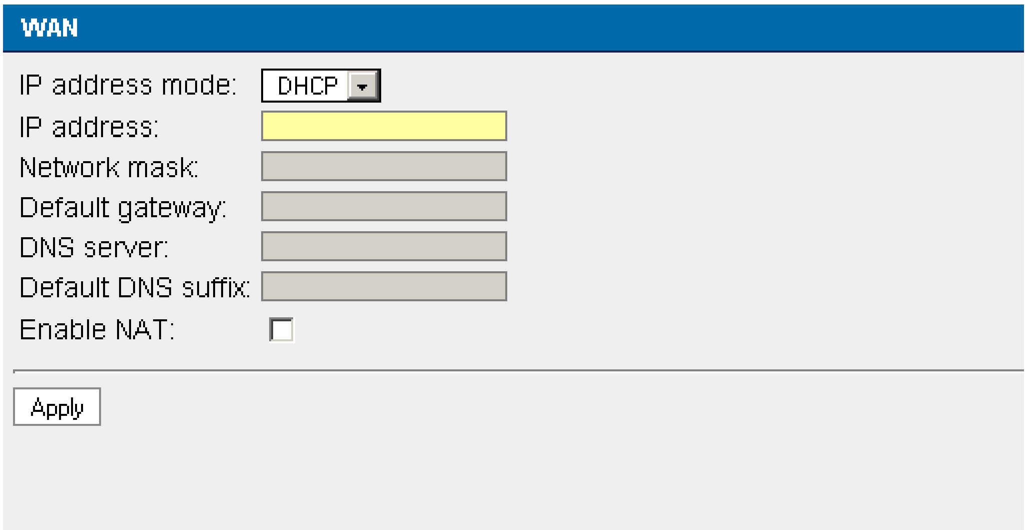
Entries made on the WAN Settings page affect the communication between the XR1000, the DSL Service Provider, and the Internet Service Provider.

Generally, entries are made on the WAN Settings page during initial setup. Once the XR1000 is successfully set up and functioning there is rarely a need to change any of the WAN settings.

WAN stands for Wide-Area Network and refers to a communication network that extends over great distances, such as countries or continents. The WWW is the classic example of a WAN.

To configure WAN settings:

1. Log-in to the XR1000 Configurator (see “Log-in and Overview of the XR1000 Configurator” on page 7).
2. On the navigation pane, click **WAN**. The WAN Configuration screen is displayed.



WAN

IP address mode: DHCP

IP address:

Network mask:

Default gateway:

DNS server:

Default DNS suffix:

Enable NAT: □

3. From the dropdown menu, choose either:

   DHCP
   The XR1000 receives its IP address automatically from your network DHCP Server. The other fields are unavailable. You do not need to make any additional entries.

Dynamic Host Configuration Protocol (DHCP) is software that automatically assigns IP addresses to client stations logging-on to a TCP/IP network.

Static
For a static or fixed IP address, configure the following parameters:

- IP address
  Enter the IP address assigned to the XR1000.
- Network mask
  Enter the subnet mask assigned to the network to which the XR1000 belongs.
- Default gateway
  Enter the IP address of the default router used by the network to which the XR1000 belongs.
- DNS Server
  Enter the IP address of the Domain Name System (DNS) server. The DNS server maintains a list of domain names and their corresponding IP addresses, and executes the substitution.
- Default DNS suffix
  Enter the suffix that is automatically added to a resolver query if it contains a host name without dots. For example, if the DNS default suffix is “mydomain.com” and the XR1000 should find host “abc” then the DNS query will be performed for name “abc.mydomain.com”

None
The WAN interface is disabled. The other fields are unavailable. You do not need to make any additional entries.

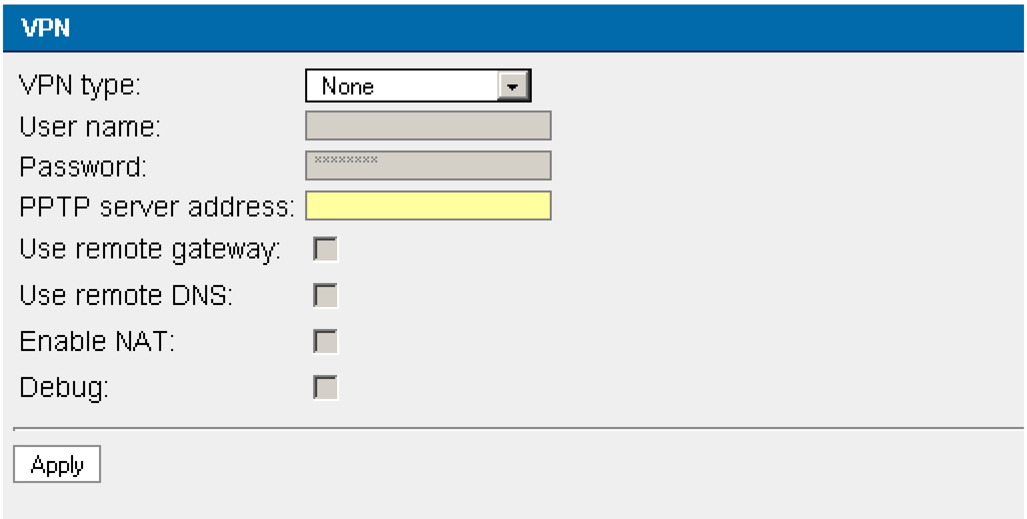
4. Select or clear the **Enable NAT** checkbox. When selected, NAT is enabled between the WAN and LAN interfaces.
5. Click **Apply**.
6. To permanently save the changes, click **Save Configuration**.

VPN Settings

A Virtual Private Network (VPN) is a smaller, independently operated network functioning within a large public network. VPNs are widely used to enable mobile and remote users to connect to their company's internal LANs. In many cases you also will need to configure the VPN in order to connect to ADSL or cable modem.

To configure the VPN:

1. Log-in to the XR1000 Configurator (see “Log-in and Overview of the XR1000 Configurator” on page 7).
2. On the navigation pane, click **VPN**. The VPN screen is displayed.



VPN

VPN type: None

User name:

Password: *********

PPTP server address:

Use remote gateway: □

Use remote DNS: □

Enable NAT: □

Debug: □

3. From the dropdown menu, select a VPN connection type:

   Point-to-Point Protocol over Ethernet (PPPoE)
   This standard is used for incorporating the Point-to-Point Protocol (PPP) into a DSL modem connection that uses Ethernet as its transport to the carrier's facilities. Used by a large number of cable modem providers, PPPoE supports the protocol layers and authentication widely used in PPP and enables a point-to-point connection to be established in the multipoint architecture of Ethernet.

   Point-to-Point Tunneling Protocol (PPTP)
   PPTP is an enhanced PPP protocol where data packets are encapsulated inside other packets. PPTP is used to transmit data in one protocol across a network that uses a different protocol and to create a virtual private network (VPN) over the Internet. Because the Internet is essentially an open network, PPTP is used to ensure that messages transmitted from one VPN node to another are secure.

   PPTP with MPPE
   Microsoft Point-to-Point Encryption (MPPE) is an encryption method from Microsoft that is used to secure VPN transmissions.

4. Configure the following parameters:

   User name
   Enter the name used to identify XR1000 to the ISP.

   Password
   Enter the password used to identify XR1000 to the ISP.

   PPTP Server address (Required for PPTP only)
   Enter the IP address of the ISP’s Gateway.

5. Select or clear the following checkboxes:

   Use remote gateway
   When selected, the IP gateway address received during establishment of the VPN connection becomes the XR1000's default IP gateway.

   Use remote DNS
   When selected, the XR1000 uses the address of the DNS server received when establishing the VPN connection.

   Enable NAT
   When selected, the XR1000 will perform NAT between the VPN network interface and the LAN.

   Debug
   When selected, the detailed diagnostic messages will be sent to the system log. Log entries can be viewed on the View Logs screen (see “View Logs” on page 26).

6. To permanently save the changes, click **Save Configuration**.

LAN Settings

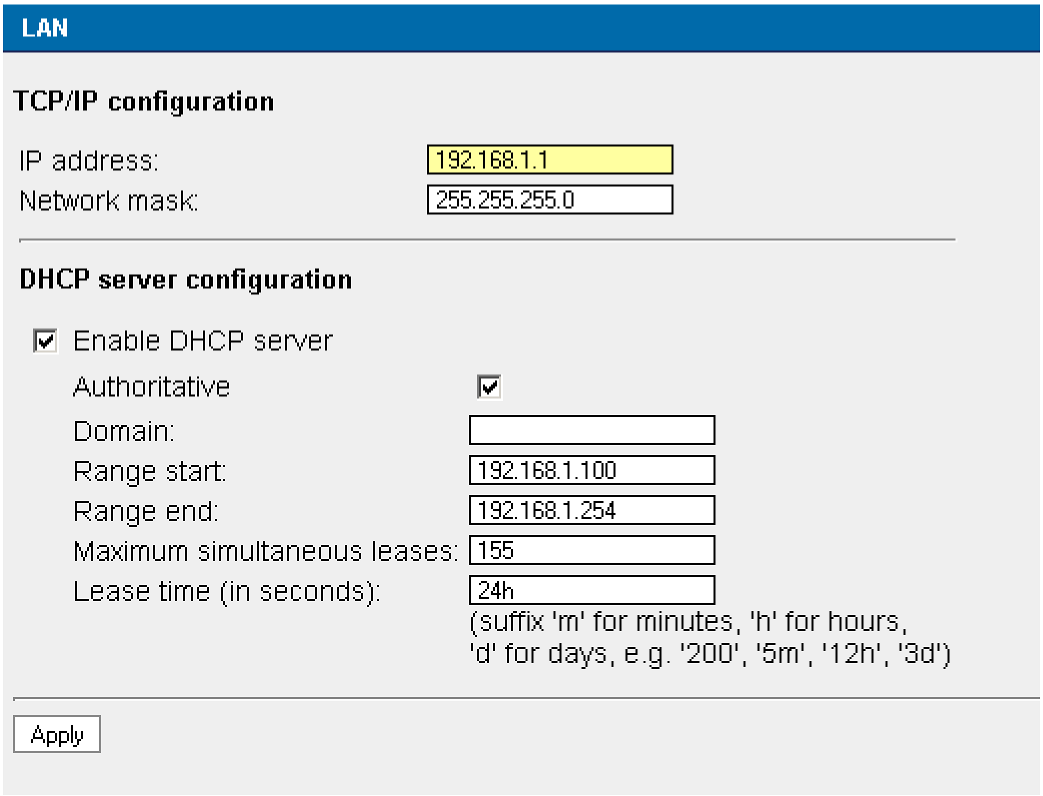
Entries made on the LAN Settings page affect the communication between the XR1000 and the computers attached to it. If you don't use the XR1000 as an IP router—that is, no computers are attached to it— then you don't need to configure the LAN interface.

Generally, entries are made on the LAN Settings page during initial setup.   Once the XR1000 is functioning, there is rarely a need to change any of the LAN settings.

If a change is made that affects the IP addresses assigned to computers connected to the XR1000 LAN port, the computers themselves must be restarted.

To configure LAN settings:

1. Ensure that you have accurate LAN settings.   Most of the parameters necessary for configuration must be assigned by the LAN Administrator.
2. Log-in to the XR1000 Configurator (see "Log-in and Overview of the XR1000 Configurator" on page 7).
3. On the navigation pane, click **LAN**.



LAN

TCP/IP configuration

IP address: 192.168.1.1
Network mask: 255.255.255.0

DHCP server configuration

☑ Enable DHCP server
Authoritative ☑
Domain:
Range start: 192.168.1.100
Range end: 192.168.1.254
Maximum simultaneous leases: 155
Lease time (in seconds): 24h
(suffix 'm' for minutes, 'h' for hours, 'd' for days, e.g. '200', '5m', '12h', '3d')

4. Define the IP address and the subnet mask used for the XR1000 LAN interface. This address may be used, for example, by the computers attached to XR1000 to connect to the XR1000 Router.

5. Select or clear the **Enable DHCP Server** checkbox. Dynamic Host Configuration Protocol (DHCP) is software that automatically assigns IP addresses to client stations logging onto a TCP/IP network. XR1000 has a DHCP server and can assign IP addresses to the computers connected to its LAN interface.

XR1000's DHCP server can be enabled or disabled. If disabled, each computer must be assigned an IP address manually.

When selected, enter the following parameters:

Authoritative
A client computer can request the last-known IP address. If the address is a valid address for the LAN network, the DHCP server can grant the request. If the address is invalid and Authoritative is enabled, the DHCP server will immediately deny the request and the client will immediately ask for a new address. If Authoritative is disabled, the server will ignore the request. The client computer will ask for a new address only after its time out.

Domain
Optionally, define a domain name for the DHCP server. Once defined, the DHCP server will return that domain to the host computers that request it. Also, this defines the domain name which is legal for hosts to claim.

Range start
This is the first IP address the XR1000 will assign. Subsequent addresses are assigned consecutively.

Range end
This is the final IP address the XR1000 will assign.

Maximum simultaneously leases
This is the maximum number of DHCP leases that the XR1000's DHCP server will support at any one time. This prevents DoS (Denial Of Service) attacks from computers that generate a large number of leases and cause the DHCP server to use too much memory. The default value is 150.

Lease time
This is the number of days an IP address assigned by the XR1000 remains unavailable even when the computer to which it was assigned does not renew the lease. For example, if Lease time is defined as two days, and a computer is removed or shut down, the IP address will become available after two days.

6. Click **Apply**.
7. To permanently save the changes, click **Save Configuration**.

Firewall

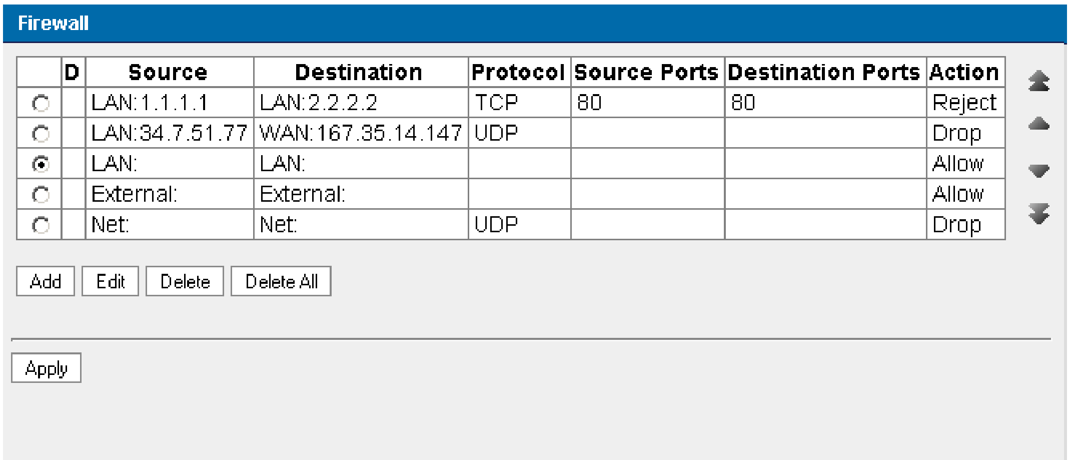
This topic defines settings which restrict access to and from the WAN, VPN, and local host (that being the XR1000 itself).

*A **Firewall** is software which disrupts free communication between trusted and untrusted networks, thereby managing the information flow and restricting access.*

Adding or Editing a Firewall Restriction

To add or edit a firewall restriction:

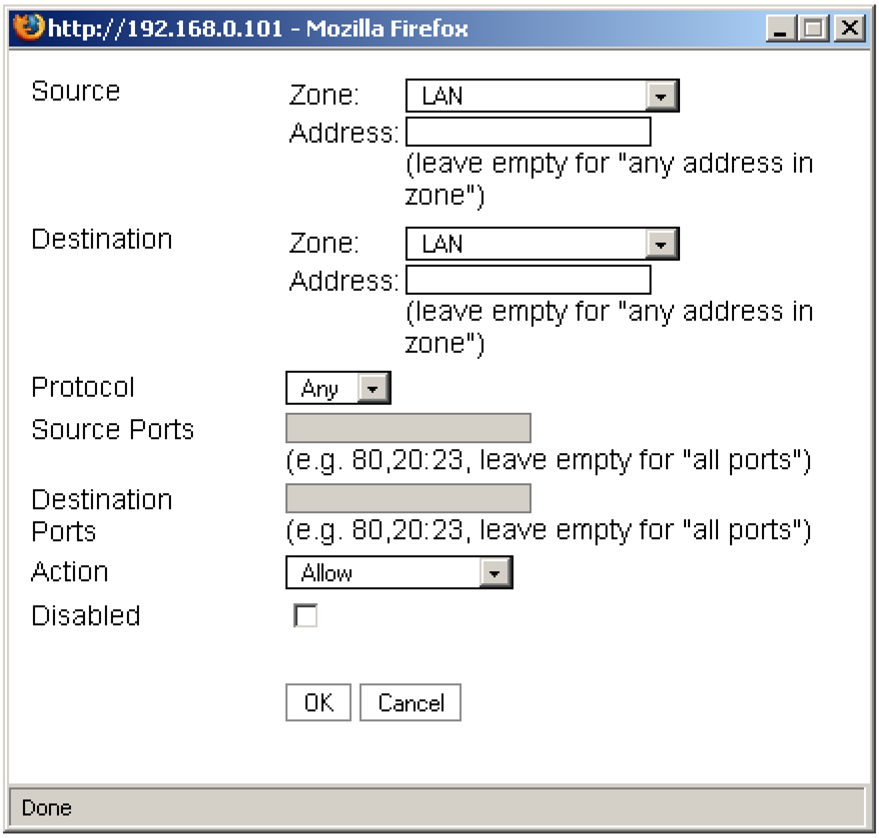
1. Log-in to the XR1000 Configurator (see “Log-in and Overview of the XR1000 Configurator” on page 7).
2. On the navigation pane, click **Firewall**. Existing Firewall rules are displayed.



Firewall

	D	Source	Destination	Protocol	Source Ports	Destination Ports	Action
○		LAN:1.1.1.1	LAN:2.2.2.2	TCP	80	80	Reject
○		LAN:34.7.51.77	WAN:167.35.14.147	UDP			Drop
●		LAN:	LAN:				Allow
○		External:	External:				Allow
○		Net:	Net:	UDP			Drop

3. On the XR1000 Configuration screen, select **Router Configurator**.
4. Click **Add**. The Firewall Settings dialog box opens.



http://192.168.0.101 - Mozilla Firefox

Source — Zone: LAN; Address: (leave empty for "any address in zone")

Destination — Zone: LAN; Address: (leave empty for "any address in zone")

Protocol: Any

Source Ports: (e.g. 80,20:23, leave empty for "all ports")

Destination Ports: (e.g. 80,20:23, leave empty for "all ports")

Action: Allow

Disabled: □

5. Define the Source connection component of the rule. The Source computer is the computer initiating the contact.

   Zone
   From the dropdown list, select the zone to which this restriction will apply. For example, if LAN is selected, the restriction will apply to all computers on the LAN.

   Address
   Optionally, enter the IP address of a host from the selected zone to which the restriction will be applied.

6. Select one of the options from the Destination address field. The Destination computer is the computer receiving the contact.

   Zone
   From the dropdown list, select the zone to which this restriction will apply. For example, if LAN is selected, the restriction will apply to all computers on the LAN.

   Address
   Optionally, enter the IP address of a host from the selected zone to which the restriction will be applied.

7. From the Protocol dropdown list, select the protocol used on the local network.

   Any
   Both TCP and UDP.

TCP (Transmission Control Protocol)
TCP is one of the two transport protocols in TCP/IP. TCP ensures that a message is sent accurately and in its entirety.

UDP (User Datagram Protocol)
UDP is the second of the two transport protocols in TCP/IP. For real-time voice and video, there is no time or reason to correct errors, and UDP is used

8. Optionally, if the protocol is defined as TCP or UDP, it is also possible to define port restrictions. If the protocol is defined as Any, it is not possible to define the ports restriction. To define port restrictions:

   a. Ensure that the protocol is defined as either TCP or UDP.

   b. Define the Source port numbers. In the common case it may be comma separated list of numbers and/or ranges. A port range may be defined as <start>:<end>. For example,

      80,82,20:23,90.

      To restrict all ports for the selected protocol, leave the Source ports field empty.

   c. Define the Destination port numbers. Rules of definition are the same as for the definition of the Source port numbers.

9. There are three types of firewall rules, distinguished by their Action. Select an Action.

   Allow
   This rule allows the connection to proceed.

   Reject
   This rule rejects the connection immediately and sends a Reject message.

   Drop
   This rule allows the connection to be dropped. No message is sent to the remote computer.

10. Click **OK**. The Firewall Settings dialog box closes.

The firewall will check the rules in an ascending order as they appear in the table.

11. Click the Move up and Move down arrows to align the rules in the order they will be applied. The firewall will check the rules in an ascending order as they appear in the table.

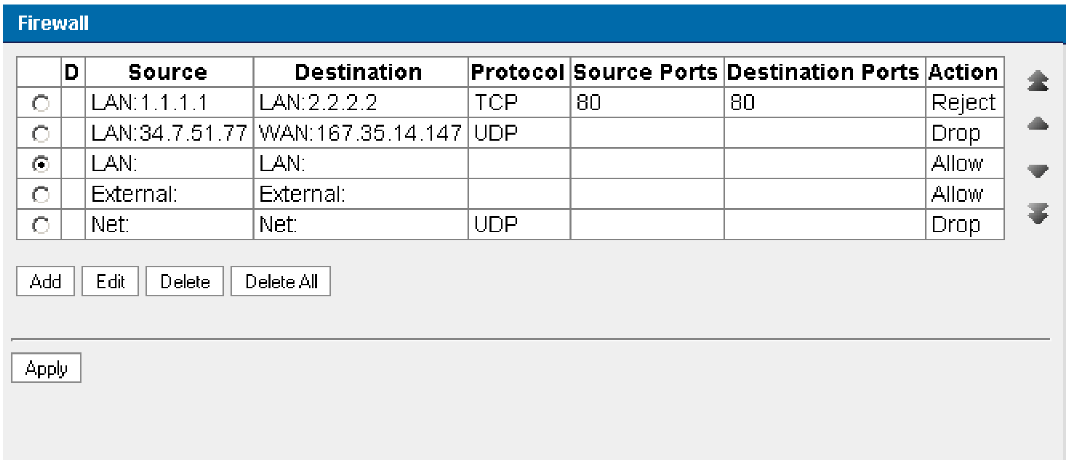
12. Click **OK** > **Apply**.

13. To permanently save the changes, click **Save Configuration**.

Removing a Firewall Restriction

To remove a firewall restriction:

1. From the Firewall screen, select the radio button beside the restriction to be removed.



Firewall

	D	Source	Destination	Protocol	Source Ports	Destination Ports	Action
○		LAN:1.1.1.1	LAN:2.2.2.2	TCP	80	80	Reject
○		LAN:34.7.51.77	WAN:167.35.14.147	UDP			Drop
●		LAN:	LAN:				Allow
○		External:	External:				Allow
○		Net:	Net:	UDP			Drop

2. Click **Delete**.
3. Click **Apply**.
4. To permanently save the changes, click **Save Configuration**.

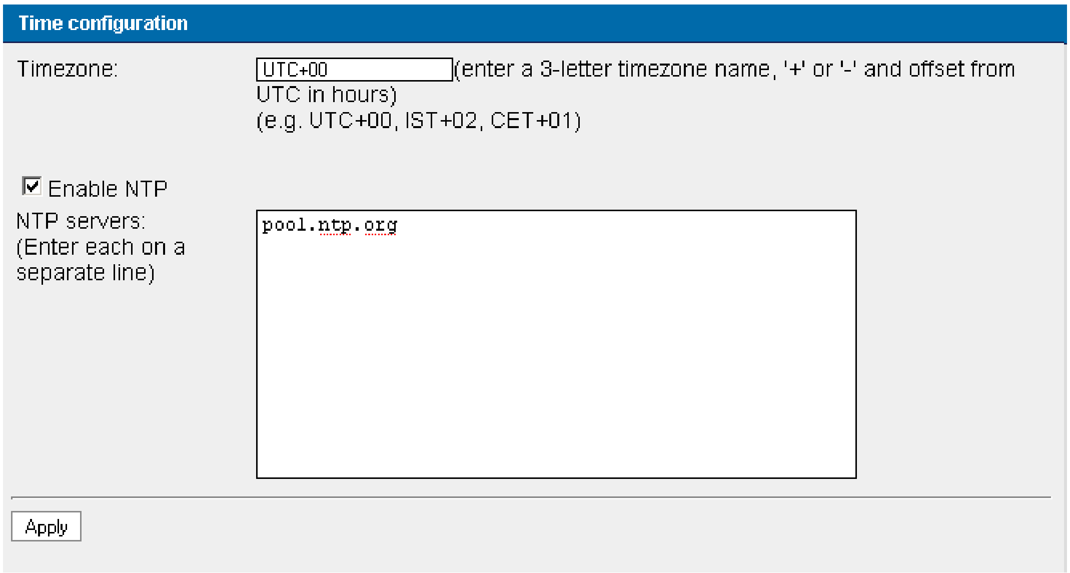
Time Zone

Time synchronization is required when using CDRs.

The XR1000 does not keep the clock over reboot. Therefore, it is recommended to enable the automatic clock synchronization via NTP protocol.

To configure the time:

1. Log-in to the XR1000 Configurator (see “Log-in and Overview of the XR1000 Configurator” on page 7).
2. On the navigation pane, click **Time Configuration**.



Time configuration

Timezone: UTC+00 (enter a 3-letter timezone name, '+' or '-' and offset from UTC in hours)
(e.g. UTC+00, IST+02, CET+01)

☑ Enable NTP

NTP servers:
(Enter each on a separate line)

```
pool.ntp.org
```

3. Define the time zone of the location where the XR1000 is installed. It should be defined as a 3-letter time zone name with an offset from the UTC time in hours. For example, if the XR1000 is located in London, then the time zone may be defined either as UTC+00 or CET-1.
4. Optionally, select the enable NTP (Network Time Protocol) checkbox and enter the DNS name or IP address of one or more NTP servers. Enter only one server per line.
5. Click **Apply**. The changes are applied only after a reboot.
6. To permanently save the changes, click **Save Configuration**.

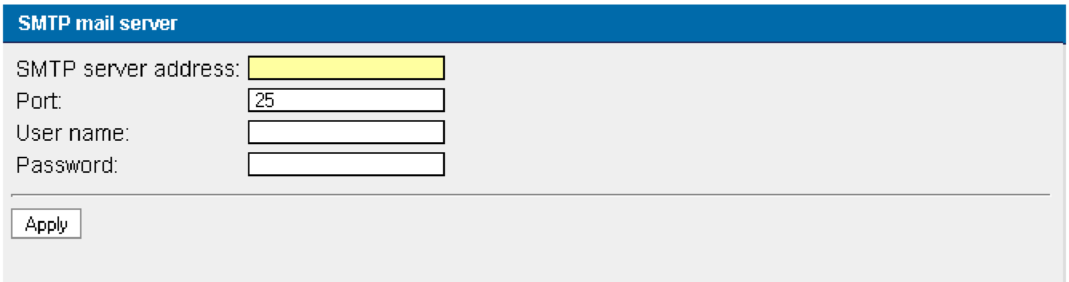
SMTP Mail Server

Use this procedure to enable the PBX to send voice mail messages as email attachments to the PBX users. You have to configure the connection to an SMTP server.

SMTP severs which require TLS (Transport Layer Security) are currently not supported.

To configure the connection to an SMTP server:

1. Log-in to the XR1000 Configurator (see “Log-in and Overview of the XR1000 Configurator” on page 7).
2. On the navigation pane, click **SMTP**.



SMTP mail server

SMTP server address:

Port: 25

User name:

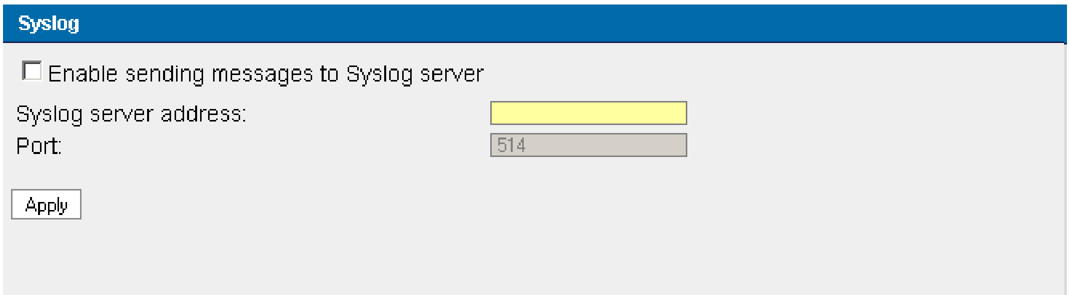
Password:

3. Enter the IP address or DNS host name of the SMTP server.
4. Enter the port number of the server. The default number is 25.
5. Enter the User name and Password required by some SMTP servers for authentication.
6. Click **Apply**.
7. To permanently save the changes, click **Save Configuration**.

Syslog

To configure XR1000 to send diagnostic messages to a syslog server:

1. Log-in to the XR1000 Configurator (see “Log-in and Overview of the XR1000 Configurator” on page 7).
2. On the navigation pane, click **Syslog**. The Syslog screen is displayed.



Syslog

□ Enable sending messages to Syslog server

Syslog server address:

Port: 514

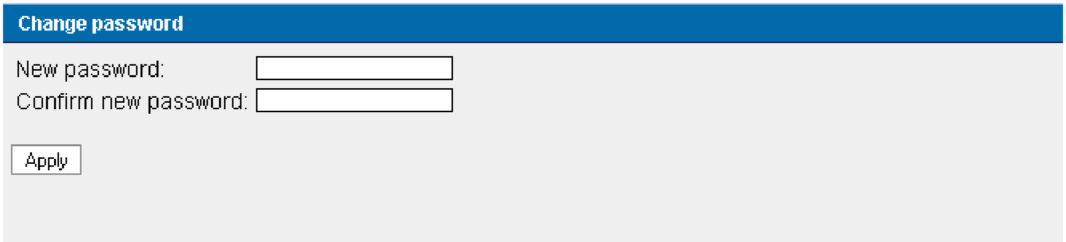
3. Select the **Enable** checkbox.
4. Enter the IP address of the Syslog server.
5. Enter the UDP port via which the Syslog server will receive messages.
6. Click **Apply**.
7. To permanently save the changes, click **Save Configuration**.

Change Password

To change the XR1000 Configurator password:

Note: *This also changes the password of the Linux "root" user.*

1. Log-in to the XR1000 Configurator (see "Log-in and Overview of the XR1000 Configurator" on page 7).
2. On the navigation pane, click **Change password**. The Change password screen is displayed.



Change password

New password:

Confirm new password:

3. Enter the new password.
4. Confirm the new password by re-entering it.
5. Click **Apply**.
6. To permanently save the changes, click **Save Configuration**.

SECTION 3:

THE MAINTENANCE MENU

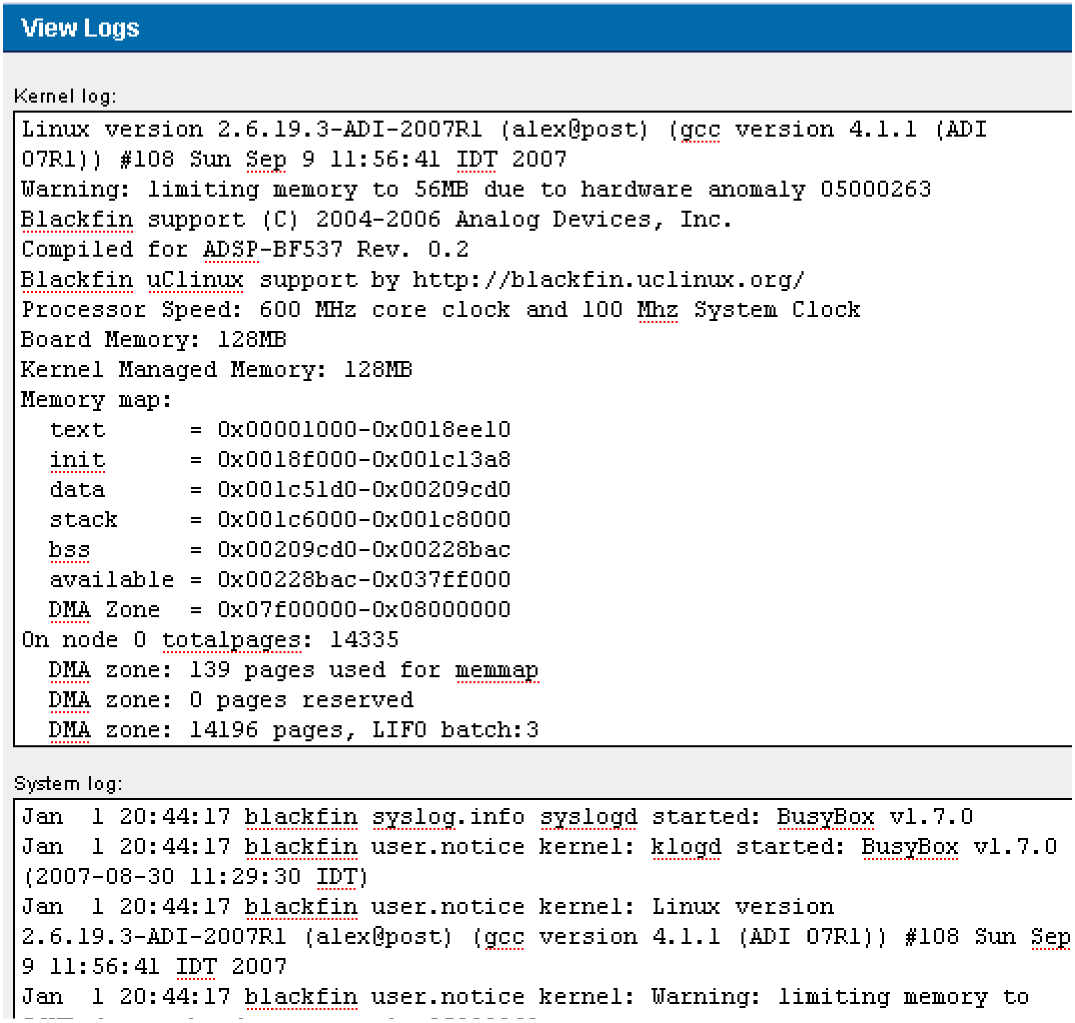
The Maintenance menu includes the following:

- “View Logs” on page 26
- “Backup and Restore” on page 27
- “Tune FXO” on page 28
- “CDR” on page 30
- “System Cleanup” on page 30
- “Software Update” on page 31
- “Remote Support” on page 32
- “Restart the Server” on page 32

View Logs

To view Kernel and System logs:

1. Log-in to the XR1000 Configurator (see “Log-in and Overview of the XR1000 Configurator” on page 7).
2. On the navigation pane, click **View Logs**.



Kernel log:

```
Linux version 2.6.19.3-ADI-2007R1 (alex@post) (gcc version 4.1.1 (ADI
07R1)) #108 Sun Sep 9 11:56:41 IDT 2007
Warning: limiting memory to 56MB due to hardware anomaly 05000263
Blackfin support (C) 2004-2006 Analog Devices, Inc.
Compiled for ADSP-BF537 Rev. 0.2
Blackfin uClinux support by http://blackfin.uclinux.org/
Processor Speed: 600 MHz core clock and 100 Mhz System Clock
Board Memory: 128MB
Kernel Managed Memory: 128MB
Memory map:
  text      = 0x00001000-0x0018ee10
  init      = 0x0018f000-0x001c13a8
  data      = 0x001c51d0-0x00209cd0
  stack     = 0x001c6000-0x001c8000
  bss       = 0x00209cd0-0x00228bac
  available = 0x00228bac-0x037ff000
  DMA Zone  = 0x07f00000-0x08000000
On node 0 totalpages: 14335
  DMA zone: 139 pages used for memmap
  DMA zone: 0 pages reserved
  DMA zone: 14196 pages, LIFO batch:3
```

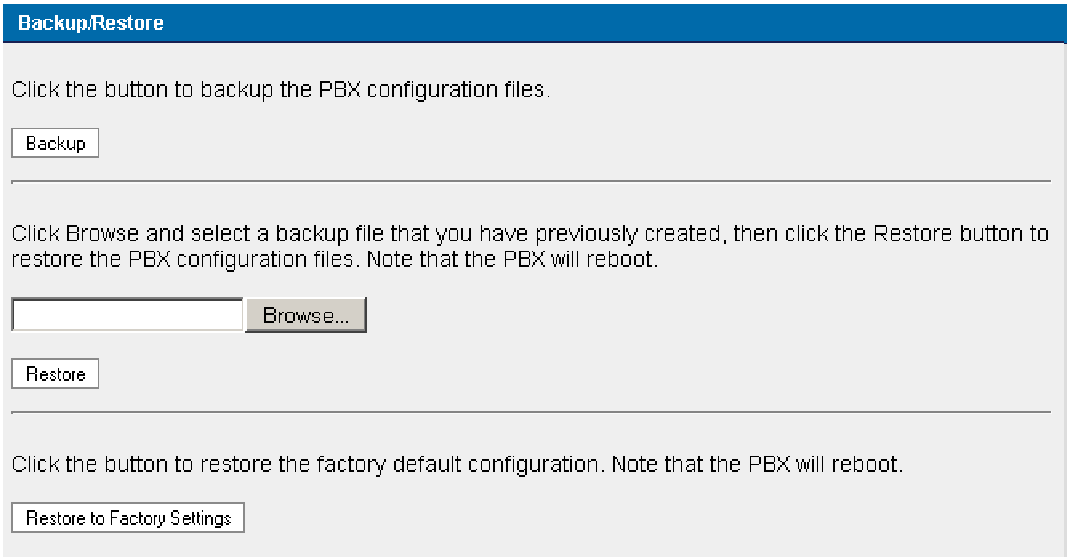
System log:

```
Jan  1 20:44:17 blackfin syslog.info syslogd started: BusyBox v1.7.0
Jan  1 20:44:17 blackfin user.notice kernel: klogd started: BusyBox v1.7.0
(2007-08-30 11:29:30 IDT)
Jan  1 20:44:17 blackfin user.notice kernel: Linux version
2.6.19.3-ADI-2007R1 (alex@post) (gcc version 4.1.1 (ADI 07R1)) #108 Sun Sep
9 11:56:41 IDT 2007
Jan  1 20:44:17 blackfin user.notice kernel: Warning: limiting memory to
```

Backup and Restore

To backup or restore the configuration of an XR1000:

1. Log-in to the XR1000 Configurator (see "Log-in and Overview of the XR1000 Configurator" on page 7).
2. On the navigation pane, click **Backup/Restore**. The Backup/Restore screen is displayed.



Backup/Restore

Click the button to backup the PBX configuration files.

Click Browse and select a backup file that you have previously created, then click the Restore button to restore the PBX configuration files. Note that the PBX will reboot.

Click the button to restore the factory default configuration. Note that the PBX will reboot.

3. Select from the following:

   To ***backup*** a configuration file:

   a. Click **Backup**. A standard **Save as** dialog box is displayed.

   b. Navigate to the location where the backup file—named config.tar.gz—is to be saved.

   c. Click **OK**. The configuration information is taken from the XR1000 and saved as a local backup file.

   To ***restore*** a configuration that had previously been saved:

   a. Click **Browse**.

   b. Locate and select the configuration file.

   c. Click **Restore**. The file will be uploaded to the XR1000, the configuration will be applied, and the unit will reboot.

Click Restore Factory Defaults to overwrite all customized parameters and return the XR1000's configuration to the original configuration assigned at the factory. Depending on the circumstances, it may be advisable to backup the configuration before restoring the factory defaults.

Tune FXO

Use Tune FXO to improve the voice quality by reducing the echo generated in any digital to analog telephony conversion. The tuning is performed using a standard Zaptel fxotune utility that runs on the XR1000.

The utility ceases the dial tone by dialing a single number—called the Dial String—and then generating tones in different frequencies while measuring the echo produced by the line. During the test, different sets of impedance parameters are used. The set that provides the best results—that is, the lowest echo—is chosen for the port.

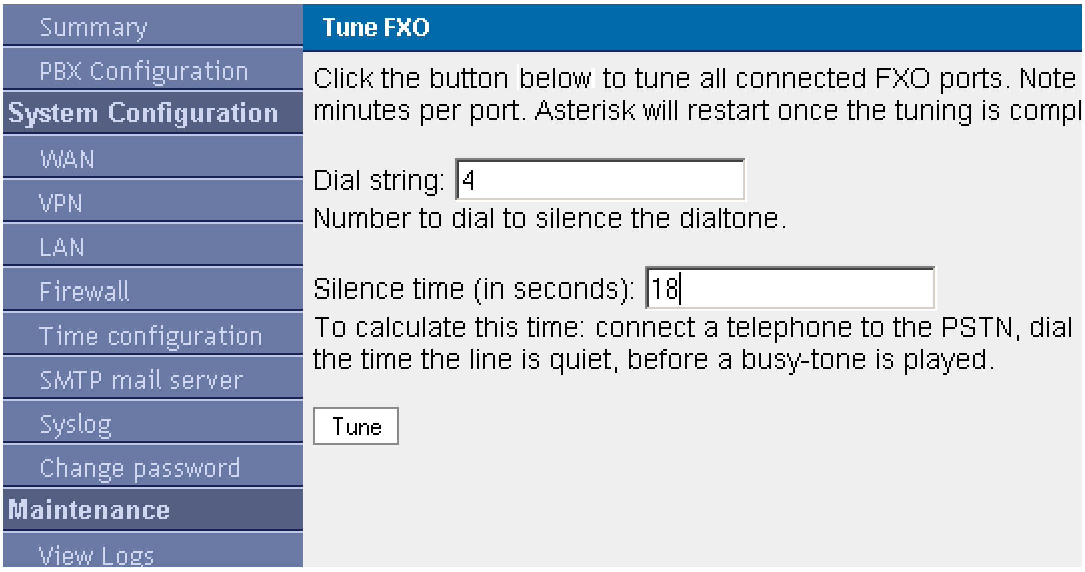
A PSTN allows a certain amount of time between the dialing of the first number and the second number. Most PSTNs will generate a busy tone if the second number is not dialed a few seconds after dialing the first number. You can measure this time by picking up the phone, dialing one number, and measuring the time before the PSTN generates the busy tone.

The parameter **Silence time** defines the measured time. The default value, 18 seconds, is normally adequate. If, however, the PSTN sends a busy tone after a time shorter than the programmed time, the tuning will not complete successfully.

⚠ ***WARNING!*** *When connecting the XR1000 to PSTN lines, ensure that a grounding tab is properly affixed (see p. 3).*

To perform the tuning:

1. Log-in to the XR1000 Configurator (see “Log-in and Overview of the XR1000 Configurator” on page 7).
2. On the navigation pane, click **Tune FXO**. The Tune FXO screen is displayed.



Tune FXO

Click the button below to tune all connected FXO ports. Note minutes per port. Asterisk will restart once the tuning is compl

Dial string: 4

Number to dial to silence the dialtone.

Silence time (in seconds): 18

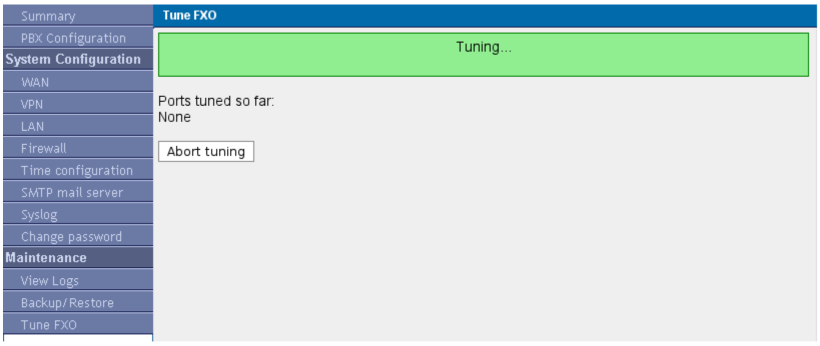
To calculate this time: connect a telephone to the PSTN, dial the time the line is quiet, before a busy-tone is played.

3. Ensure that a grounding tab is properly affixed (see p. 3).

4. Ensure that all used PSTN lines are properly connected to the corresponding FXO connectors.

Note: *If you switch lines between connectors, you must rerun FXO Tune.*

5. Define the **Dial string**. This is the number the utility dials to cease the dial tone.
6. Define the **Silence time**. This is the amount of time the PSTN will allow between the dialing of the first number and the second number.
7. Click **Tune**.

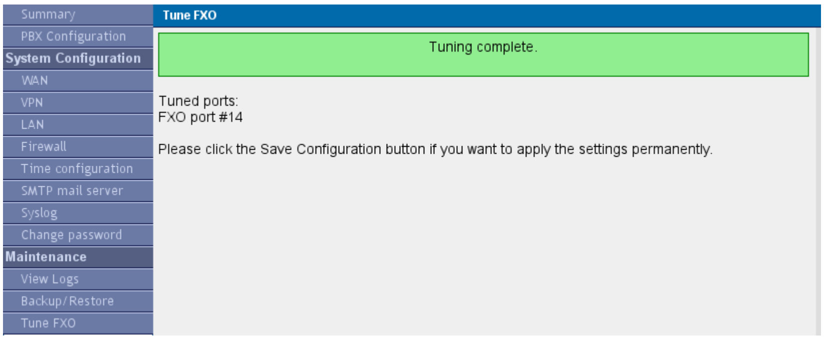


Tune FXO

Tuning...

Ports tuned so far:
None

The script detects the FXO ports where the PSTN lines are connected and then runs the fxotune utility to adjust those ports' parameters. The procedure may take several minutes.



Tune FXO

Tuning complete.

Tuned ports:
FXO port #14

Please click the Save Configuration button if you want to apply the settings permanently.

8. To permanently save the configuration, click **Save Configuration**.

CDR

Call Detail Records (CDR) keep track of a XR1000's incoming and outgoing calls. Starting from XR1000 software version 0.95, CDR collecting is disabled by default. See "CDR Reader" on page 68 for further details.

To review the CDRs, they must first be downloaded.

The records are presented in the comma-separated values (CSV) format.

1. Log-in to the XR1000 Configurator (see "Log-in and Overview of the XR1000 Configurator" on page 7).
2. On the navigation pane, click **CDR**. The CDR screen is displayed.
3. Click **Download**.
4. Navigate to the location where the CDR file—named Master.csv—is to be saved.
5. Click **OK**. The CDRs are taken from the XR1000 and saved as a local file.
6. Optionally, click Delete CDRs to delete the records from the XR1000.

System Cleanup

Use System Cleanup to delete temporary files from the flash drive. Log files, CDRs, and voice mail can be deleted.

1. Log-in to the XR1000 Configurator (see "Log-in and Overview of the XR1000 Configurator" on page 7).
2. On the navigation pane, click **System cleanup**. The System cleanup screen is displayed.
3. Select or clear the checkboxes.

There is no confirmation message.

4. Click **Cleanup**.

Software Update

Use the Software Update page to install the latest version of the XR1000 software.

Note: *A firmware upgrade does not cause lose of your device configuration. Nevertheless, we recommend that you back-up the configuration before updating.*

1. Download the appropriate firmware version file and save it in a folder on your computer. Versions can be downloaded from: http://www.xorcom.com/products/xr1000/xr1000_firmware
2. Reboot the XR1000 (see "Restart the Server" on page 32).
3. Log-in to the XR1000 Configurator (see "Log-in and Overview of the XR1000 Configurator" on page 7).
4. On the navigation pane, click **Software update**. The Software update screen is displayed.
5. Click Browse.
6. Locate and select the configuration file. The firmware file name format is:

   `xr1000-V.VV-YYYYMMDDhhmm.tar.gz`

   where,

   `V.VV` = version number

   `YYYYMMDDhhmm` = year, month, day, hour and minutes when the package was built.
7. Click **Update**.

   The update process takes several minutes and may take more time if the update is performed via the Internet.

   When the version update is completed, the device reboots itself and the LEDs flash. However, there is no message on this page indicating that the update is complete.

Remote Support

Use this screen to enable a Technical Support engineer to create a RapidTunneling ™ remote SSH connection to your PBX. The connection is possible even when your device is located behind a NAT router and usually it does not require any changes in the router configuration.

A Technical Support engineer will prepare and send you by email a file that contains address information and a public encryption key needed for establishing secure communication channel from your PBX to a special Technical Support Server. Once the connection is established, the engineer will create an SSH tunnel and access your PBX through that server.

Note: *Your unit will be available for remote access only through that special server and only until either the engineer disconnects your PBX from the server or until you reboot the PBX.*

In order to enable RapidTunneling ™ communication:

1. Save onto your computer the file delivered to you from Xorcom Technical Support.
2. Connect to the unit with your Internet browser and log in to the system.
3. From the navigation pane, select **Remote Support**.
4. Click **Browse**.
5. Locate and select the file that you received from Xorcom Technical Support.
6. Click **Connect**. This enables remote connection to your PBX.

To break down the RapidTunneling ™ communication tunnel:

1. Connect to the unit with your Internet browser and log in to the system.
2. From the navigation pane, select **Restart**.
3. Click **Reboot System**. This breaks down the remote connection to your PBX.

Restart the Server

Use the Restart screen to restart either the Asterisk PBX software only or to reboot the entire XR1000 system.

SECTION 4:

PBX CONFIGURATION

Use the XR1000 browser-based PBX Configurator to configure the dial plan features.

This section includes:

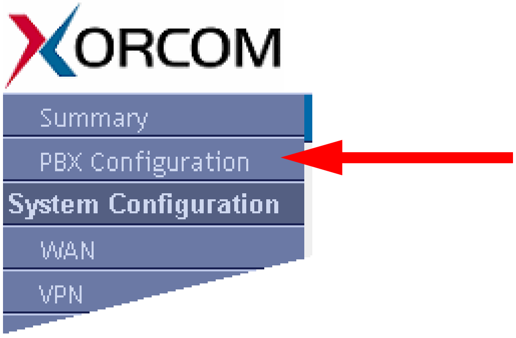
Overview

Configure PBX functionality using the modified Asterisk GUI supplied with the XR1000. Access PBX services from analog telephones using the special phone numbers that appear in “Special Phone Numbers for Accessing PBX Service” on page 38.

Running the PBX Configurator

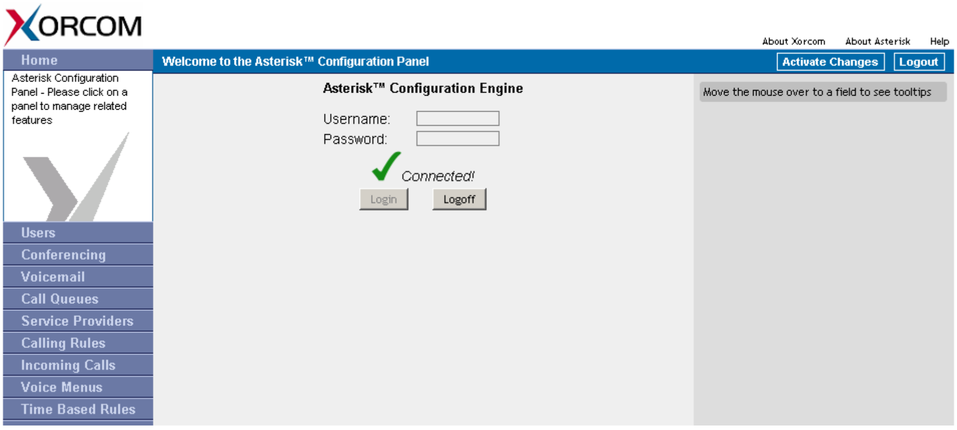
To run the PBX configurator:

1. Log-in to the XR1000 Configurator (see “Log-in and Overview of the XR1000 Configurator” on page 7).
2. On the navigation pane, click **PBX Configuration**.



The PBX Configuration Panel opens in a separate window or tab of your Internet browser.

Note: *You do not need to re-enter the Log-in name and password.*



Welcome to the Asterisk™ Configuration Panel

Asterisk Configuration Panel - Please click on a panel to manage related features

Asterisk™ Configuration Engine

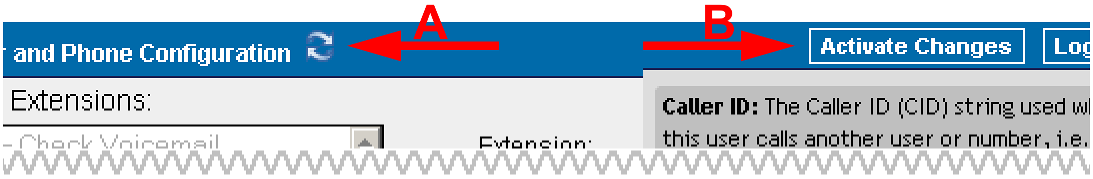
Username:

Password:

✓ Connected!

Move the mouse over to a field to see tooltips

3. We recommend becoming familiar with the following interface elements that appear in the blue title bar:



Caller ID: The Caller ID (CID) string used wl this user calls another user or number, i.e.

Element	Explanation
es ⟳	These two arrows (A above) are a Refresh button. Click here to reload the displayed information.
Activate Changes	Clicking this button—B above, located in the upper right corner—forces XR1000 to reload the configuration files. Note that you must also save the changes on the flash memory—by clicking the **Save Configuration** button on the System Configuration screen—if you want the changes to be permanent.

Principles of Dialing Plan Configuration

For the most part, PBX extensions are configured on the Users screen. There are exceptions—that is, optional extension numbers—that are defined on other screens. For example, when you add a Ring Group (see “Ring Groups” on page 64) you have the option to define a phone number that may be used for dialing to that Ring Group.

The PBX communicates with the external world via Service Providers that are defined on the Service Providers screen. Each service provider is actually either an analog (FXO) or a VoIP (SIP or IAX2) trunk.

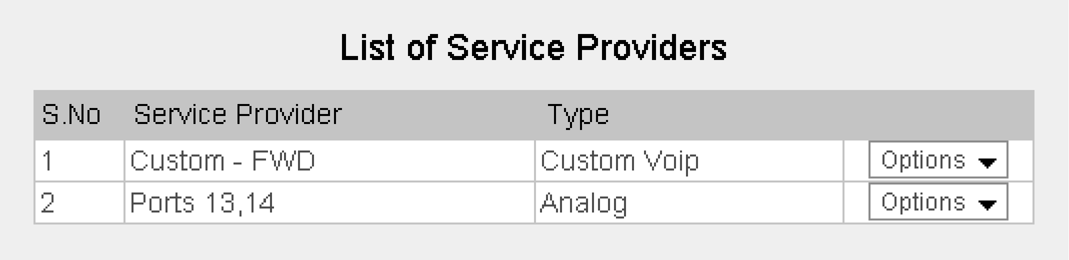
The dialing rules that determine how to dial to a Service Provider are defined on the Calling Rules screen (see “Calling Rules” on page 52). The rules are grouped in Dial Plans. Each PBX extension has a Dial Plan assigned to it. This allows the System Administrator to define different rules and permissions for outward calling for particular extensions.

In the following example, there are two Service Providers:

- FXO lines connected to the PSTN
- FreeWorld Dialup (FWD) VoIP provider.

The PBX must be configured such that users 401 and 402 will be able to make outward calls via both providers and user 403 must be able to make calls only via FWD. User 404 must be able to make internal calls only.

1. First, define both Service Providers in the Service Providers dialog:

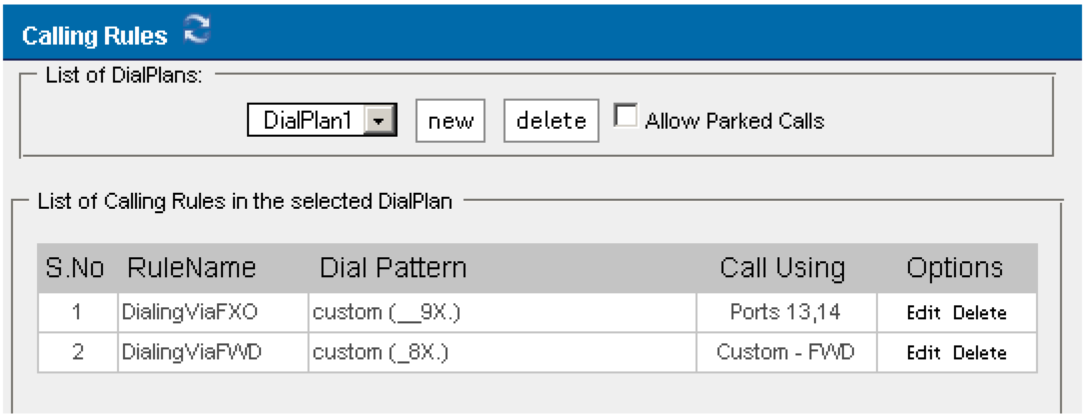


List of Service Providers

S.No	Service Provider	Type	
1	Custom - FWD	Custom Voip	Options ▾
2	Ports 13,14	Analog	Options ▾

2. Then, create three Dial Plans using the Calling Rules screen.

Dial Plan 1
The first rule requires that you dial 9 and then any number of digits in order to make call via an FXO line. The second rule requires that you dial 8 and then any number of digits in order to make call via FWD.



Calling Rules

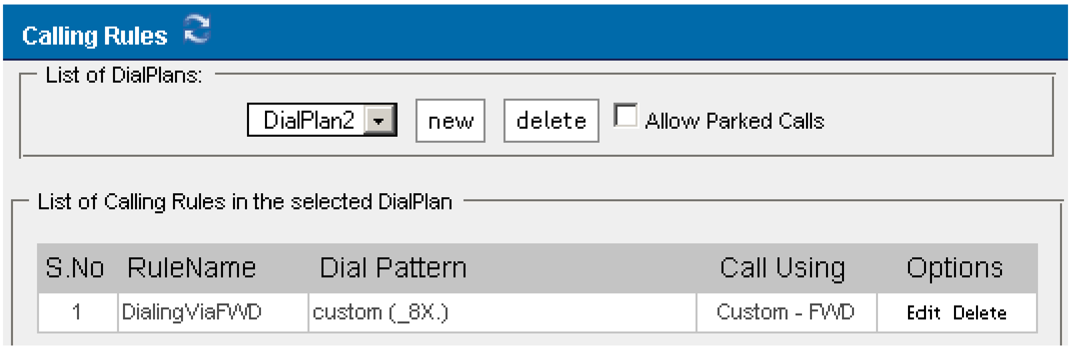
List of DialPlans: DialPlan1

☐ Allow Parked Calls

List of Calling Rules in the selected DialPlan

S.No	RuleName	Dial Pattern	Call Using	Options
1	DialingViaFXO	custom (__9X.)	Ports 13,14	Edit Delete
2	DialingViaFWD	custom (_8X.)	Custom - FWD	Edit Delete

Dial Plan 2
Dial Plan 2 has only one rule which requires that you dial 8 and then any number of digits in order to make call via FWD.



Calling Rules

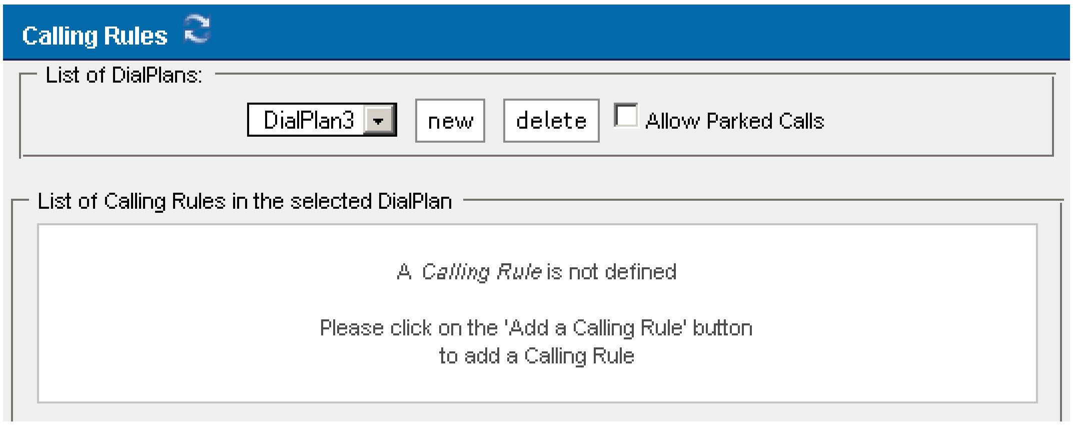
List of DialPlans: DialPlan2

☐ Allow Parked Calls

List of Calling Rules in the selected DialPlan

S.No	RuleName	Dial Pattern	Call Using	Options
1	DialingViaFWD	custom (_8X.)	Custom - FWD	Edit Delete

Dial Plan 3
This dial plan does not include any rules.



Calling Rules

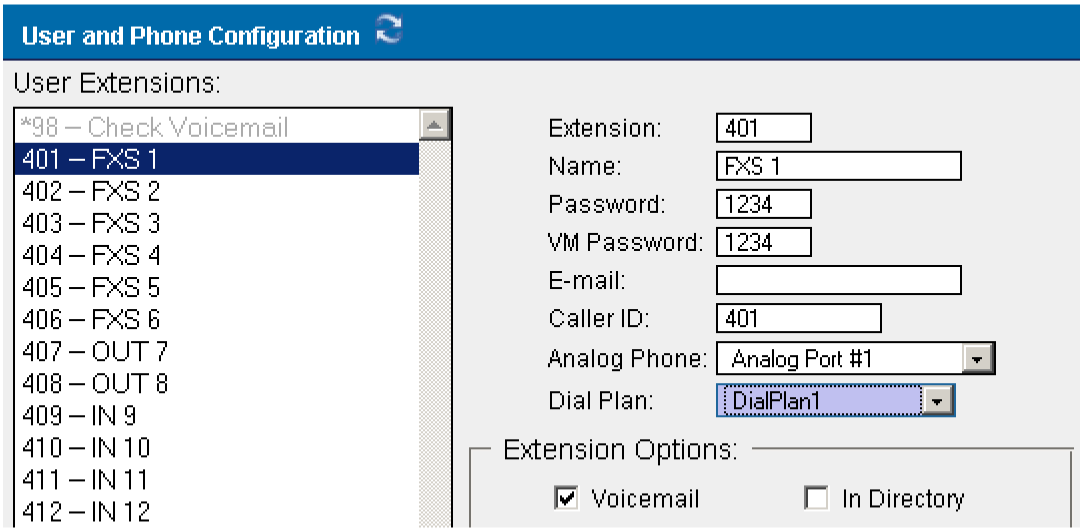
List of DialPlans: DialPlan3 new delete ☐ Allow Parked Calls

List of Calling Rules in the selected DialPlan

A *Calling Rule* is not defined

Please click on the 'Add a Calling Rule' button
to add a Calling Rule

3. Now assign the suitable Dial Plan to each user. For example, user 401 and 402 must refer to DialPlan1:



User and Phone Configuration

User Extensions:

*98 – Check Voicemail
401 – FXS 1
402 – FXS 2
403 – FXS 3
404 – FXS 4
405 – FXS 5
406 – FXS 6
407 – OUT 7
408 – OUT 8
409 – IN 9
410 – IN 10
411 – IN 11
412 – IN 12

Extension: 401
Name: FXS 1
Password: 1234
VM Password: 1234
E-mail:
Caller ID: 401
Analog Phone: Analog Port #1
Dial Plan: DialPlan1

Extension Options:

☑ Voicemail ☐ In Directory

4. User 403 must refer to DialPlan2 and user 404 to the empty DialPlan3.

Special Phone Numbers for Accessing PBX Service

The special phone numbers that can be used to access the PBX services from analog telephones are:

*65	Announces your extension number
*66	Announces the XR1000 IP address. If there is an active VPN connection, then the VPN IP address will be announced. Otherwise, the IP address of the Ethernet WAN interface will be announced.
*98	Accesses voice mail
The following services are available only for the analog FXS extensions:	
*67	Disables Caller ID
*69	Announces the last caller’s Caller ID (call return)
*70	Disables call waiting (for current call)
*72	Activates unconditional call forwarding
*73	Deactivates unconditional call forwarding
*78	Enables Do Not Disturb
*79	Disables Do Not Disturb
*60	Adds last caller’s caller ID to blacklist
*82	Enables Caller ID (only if disabled with *67)

For example, to activate the unconditional call forwarding for an FXS extension:

a. Dial *72.

b. Wait for a special tone.

c. Dial the new number that calls will be forwarded to.

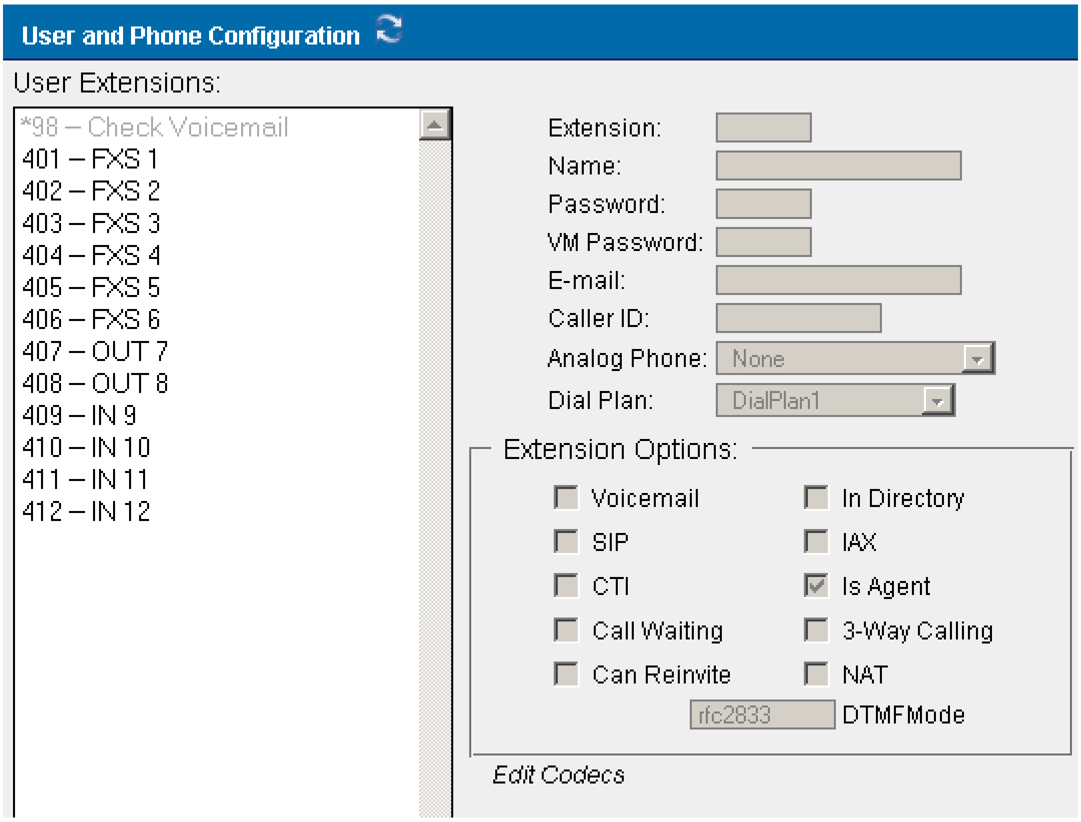
d. Wait for another tone.

e. Hang up.

Adding User Extensions

To add a User Extension:

1. Run the PBX Configurator (see “Running the PBX Configurator” on page 34).
2. On the navigation pane, click **User Extensions**. Existing extensions are displayed.



User and Phone Configuration

User Extensions:

*98 – Check Voicemail
401 – FXS 1
402 – FXS 2
403 – FXS 3
404 – FXS 4
405 – FXS 5
406 – FXS 6
407 – OUT 7
408 – OUT 8
409 – IN 9
410 – IN 10
411 – IN 11
412 – IN 12

Extension:
Name:
Password:
VM Password:
E-mail:
Caller ID:
Analog Phone: None
Dial Plan: DialPlan1

Extension Options:

☐ Voicemail ☐ In Directory
☐ SIP ☐ IAX
☐ CTI ☑ Is Agent
☐ Call Waiting ☐ 3-Way Calling
☐ Can Reinvite ☐ NAT
rfc2833 DTMFMode

Edit Codecs

3. Click **New**.
4. Define the following parameters:

Extension
Enter the extension assigned to the user.

Name
Enter the name of the individual assigned to this extension.

Note: *The Dial By Name Directory uses this information to route calls.*

Password
Enter the password used to access voice-mail for the extension.

Email Address
Enter the email address to which voice-mail received by this extension will be forwarded.

Caller ID
Enter the Caller ID displayed to the person receiving a call from this extension.

Note: *If you use a letter for the Caller ID, the Caller ID will not be displayed on an analog phone since analog phones can display numbers only.*

Analog Phone
From the dropdown menu, select the analog phone port which this extension will access.

Dial Plan
From the dropdown menu, select a Dial Plan. Use Dial Plans to configure calling permissions for each PBX extension. The Dial Plan is configured on the Calling Rules screen. Each Dial Plan can include a different set of calling rules.

5. Select or clear the following **Extension Options**:

Note: *It is possible to define simultaneously SIP, IAX and Analog Telephony capabilities for each individual user.*

Voice-mail
Select this option to create a voice-mail box for the extension.

In Directory listing
Select this option to include this user in the telephony directory. The telephony directory is referred to by the Asterisk Directory application that allows users to call the PBX extensions by dialing the three first letters of the associated name. It also is needed to configure a *Go to Directory* as an item in the Voice Menu (see “Voice Menus” on page 57).

SIP
Select this options when the extension uses the SIP protocol.

IAX
Select this options when the extension uses the IAX protocol.

CTI (Computer Telephony Integration)
Select this option to enable users to connect applications to the XR1000 Management Interface. When selected, a Asterisk Management Interface user will automatically be created using the Name and Password defined above.

Is Agent
When selected, the extension may be a member of a Call Queue (see “Configuring Call Queues” on page 46).

Call Waiting
When selected, the called user hears a signal if the extension receives a second incoming call. The caller hears the normal ring-back tone and does not get any indication that the called number is engaged. The feature is applicable for the analog PBX extensions only.

3-Way Calling

When selected, the user can dial out to a second phone number thereby creating a 3-way conference call.

Can Reinvite

This feature is only applicable if the user has SIP or/and IAX capabilities. When enabled, the Asterisk will perform IP/port re-negotiation and force the call parties to send voice packets directly between them.

NAT

NAT (Network Address Translation) should be enabled when the XR1000 is located on the open Internet and the user is located behind a NAT system. In many cases, selecting NAT helps solve the *one-way voice* problem when the SIP protocol is used.

DTMF Mode

DTMF Mode (Dual-tone multi-frequency) defines how DTMF signals are sent during SIP calls. Select one of the following:

- rfc2833
  Sends DTMFs according to the RFC-2833 (the default value)
- info
  Sends DTMFs by using SIP INFO messages
- inband
  Sends DTMFs as regular voice
- auto
  Sends DTMFs according to the RFC-2833 if the partner supports this method. If not, they are sent as inband.

6. Define the codec list to be used for VoIP calls.

Note: *Codec types are voice compression/decompression (CoDec) methods. The Codec type is negotiated with the remote VoIP device per call.*

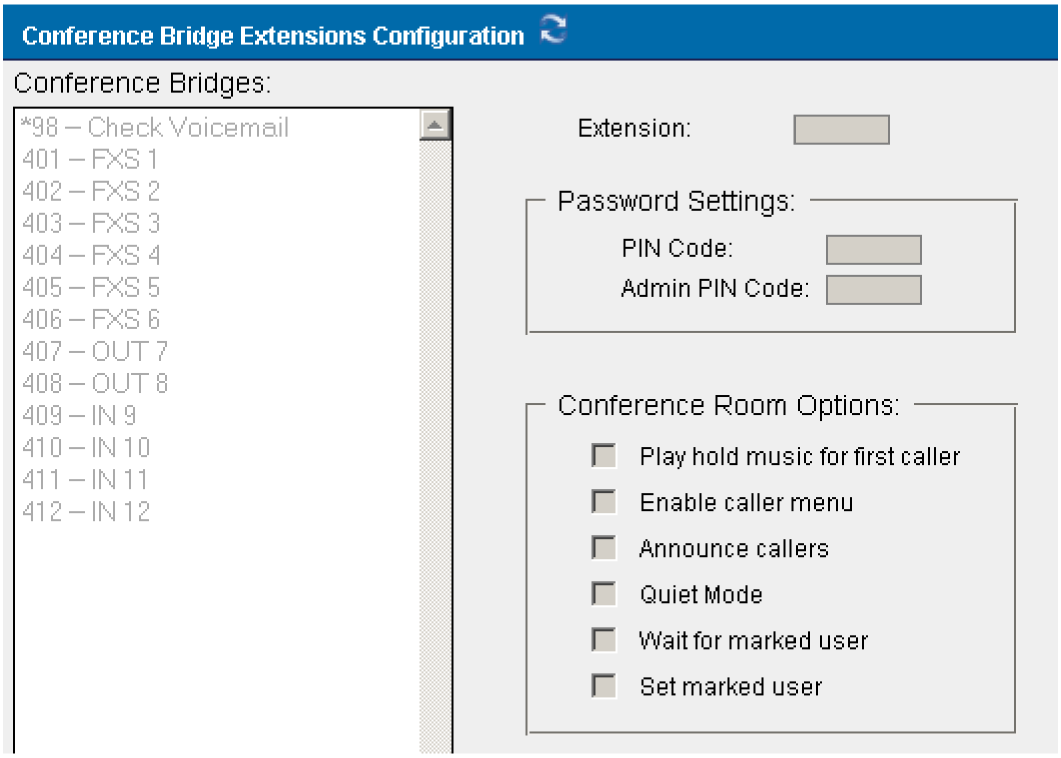
WARNING! *The XR1000 must have at least one enabled codec that is common with the remote VoIP device. Otherwise, the call will be rejected.*

7. To permanently save the changes, return to the System Configuration browser-window, and click **Save Configuration**.

Enabling Conferencing

Conferencing allows creation of phone-calls “Bridges” that include more than 3 parties. To create a new phone-call Bridge:

1. Run the PBX Configurator (see “Running the PBX Configurator” on page 34).
2. On the navigation pane, click **Conferencing**. Existing Conference Bridges are displayed.



Conference Bridge Extensions Configuration

Conference Bridges:

*98 – Check Voicemail
401 – FXS 1
402 – FXS 2
403 – FXS 3
404 – FXS 4
405 – FXS 5
406 – FXS 6
407 – OUT 7
408 – OUT 8
409 – IN 9
410 – IN 10
411 – IN 11
412 – IN 12

Extension:

Password Settings:

PIN Code:

Admin PIN Code:

Conference Room Options:

□ Play hold music for first caller
□ Enable caller menu
□ Announce callers
□ Quiet Mode
□ Wait for marked user
□ Set marked user

3. Click **New**.
4. Define the following parameters:
5. Enter the **Extension** dialed to access the Bridge.
6. Enter the **Password** settings

   PIN Code
   This is the number entered by users to participate in the conference.

   Administrator PIN Code
   This is the number entered by the moderator of the conference to open the conference bridge.

7. Select or clear the following Conference Room options:

   Play hold music for first caller
   When selected, music is played for the first caller entering the conference. The music ends when a second caller joins.

   Enable caller menu
   When selected, users can access the Conference Bridge Menu by pressing the asterisk (*) key.

Announce callers
When selected, new callers are identified when they arrive.

Quiet Mode
This feature confines participants to listen-only access. For example, by creating two access points, one group can be confined to Quiet Mode while the speaker lectures.

Wait for Marked User
Use this feature to confine all participants to Quiet Mode until a participant—called the Marked User and using a unique extension—arrives. After the Marked User arrives, the audio is activated and the other participants can speak to each other.

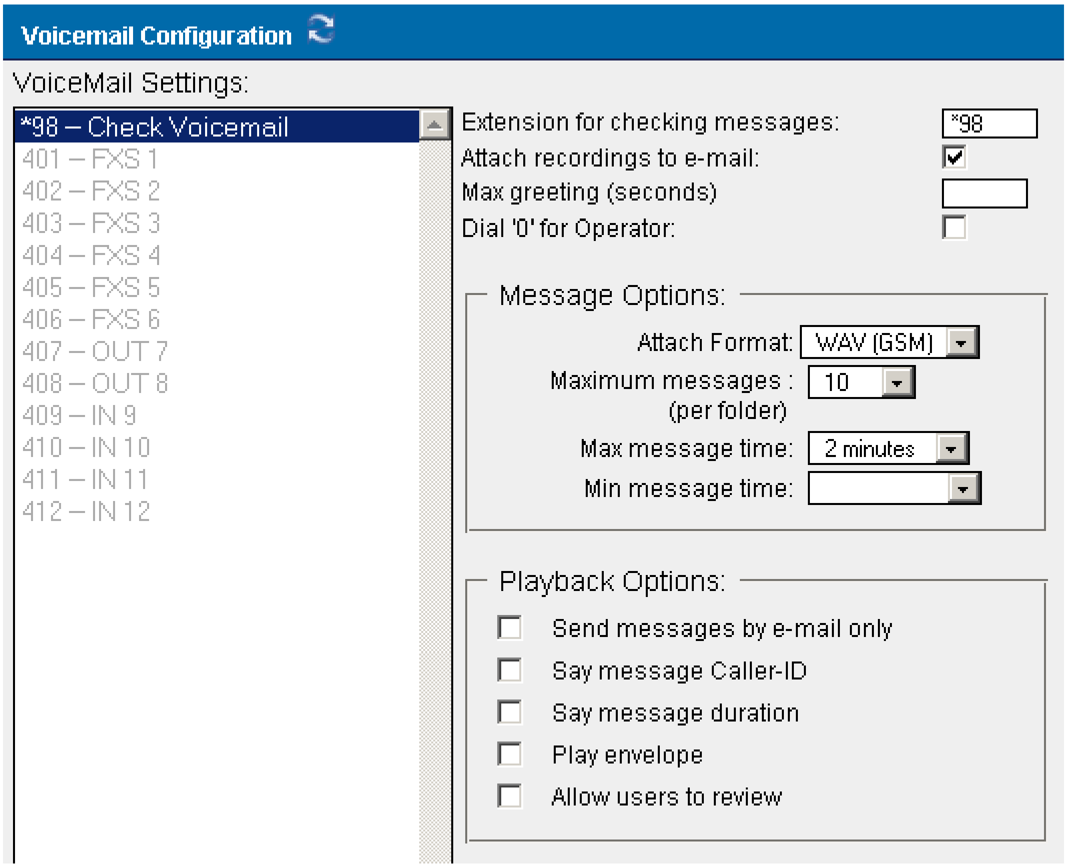
Set Marked User
Select this option to designate the Marked User’s extension.

8. Optionally, test the changes.
9. To permanently save the changes, return to the System Configuration browser-window, and click **Save Configuration**.

Configuring Voicemail

To configure voice-mail:

1. Run the PBX Configurator (see “Running the PBX Configurator” on page 34).
2. Optionally, if you will use voice-mail-to-email functionality, configure SMTP mail server parameters (see “SMTP Mail Server” on page 22).
3. On the navigation pane, click **Voicemail**. Voicemail Configuration page is displayed.



Voicemail Configuration

VoiceMail Settings:

*98 – Check Voicemail
401 – FXS 1
402 – FXS 2
403 – FXS 3
404 – FXS 4
405 – FXS 5
406 – FXS 6
407 – OUT 7
408 – OUT 8
409 – IN 9
410 – IN 10
411 – IN 11
412 – IN 12

Extension for checking messages: *98
Attach recordings to e-mail: ☑
Max greeting (seconds)
Dial '0' for Operator: □

Message Options:
Attach Format: WAV (GSM)
Maximum messages : (per folder) 10
Max message time: 2 minutes
Min message time:

Playback Options:
□ Send messages by e-mail only
□ Say message Caller-ID
□ Say message duration
□ Play envelope
□ Allow users to review

4. Enter the **Extension for checking messages**. This is the extension Users call to access their voice-mail. By default, the XR1000 uses extension *98.
5. Select or clear the **Attach recordings to email** checkbox. When selected, voice-mail is sent to the users email address.
6. Enter a **Max greeting** length. This specifies the maximum duration of the recorded voice-mail greeting.
7. Select or clear the **Dial "0" for Operator** checkbox. When selected, callers can can transfer to the operator by pressing **0**. If cleared, pressing 0 is ignored.
8. Configure the **Message Options**:

   Message Format
   From the dropdown menu, select the format in which messages are delivered by email.

Maximum Messages per Folder
Enter the maximum number of messages allowed in a voice-mail box. The default maximum is 10.

Maximum Message Time
Enter the maximum allowed duration of a message. The maximum allowed is two minutes.

Minimum Message Time
Enter the minimum allowed duration of a message. The default value is zero.

9. Select or clear the following **Playback Options**:

   Send Messages by email Only
   When selected, notification of messages is by email only. Voice messages are not saved. Make sure that the SMTP server is configured (see "SMTP Mail Server" on page 22).

   Say Message Caller-ID
   When selected, the caller ID is announced before the voice-mail message is played.

   Say Message Duration
   When selected, the duration of the message is announced before the voice-mail message is played.

   Play Envelope
   When selected, the date, time, and caller ID of the message is announced before the voice-mail message is played.

   Allow Users to Review
   When selected, callers can review and edit their message before it is saved.

10. Optionally, test the changes.

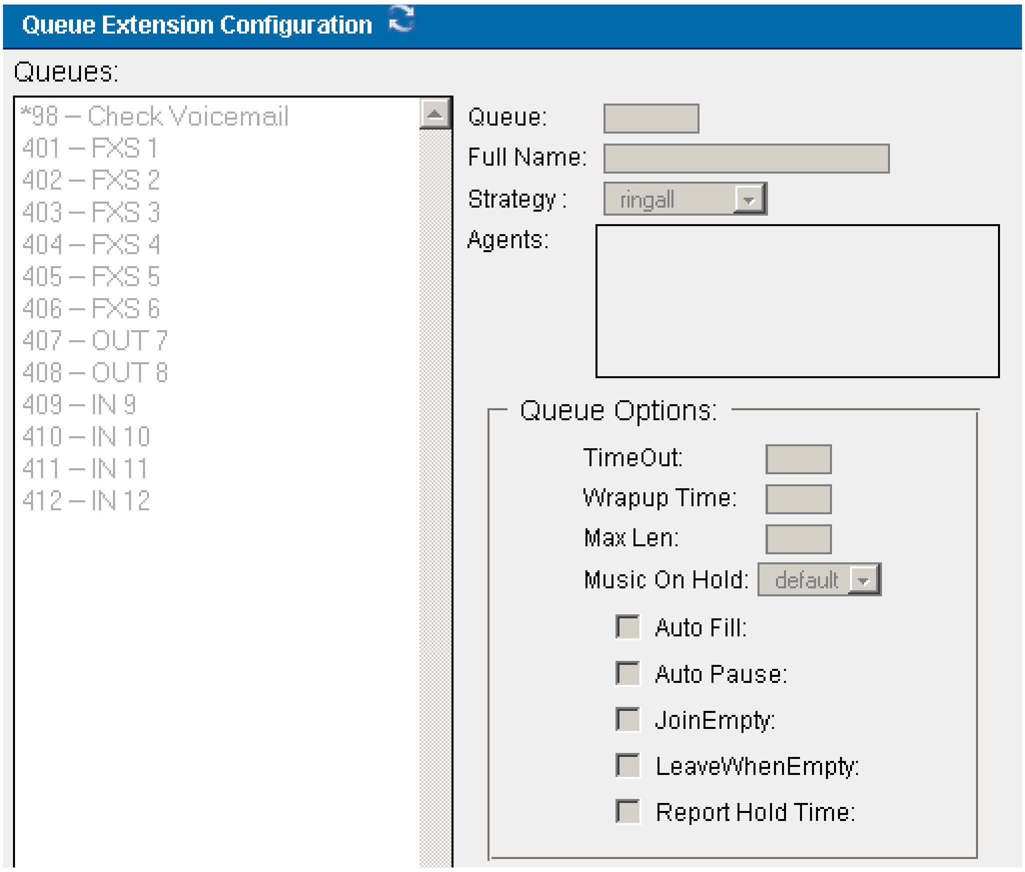
11. To permanently save the changes, return to the System Configuration browser-window, and click **Save Configuration**.

12. Enable Voicemail and define a Voicemail password for each user that will have a voice mail box (see "Adding User Extensions" on page 39).

Configuring Call Queues

A call queue stacks incoming calls in the order they are received. Calls are allocated to operators according to the queue settings. To configure a Call Queue:

1. Run the PBX Configurator (see “Running the PBX Configurator” on page 34).
2. Ensure that **Is Agent** is enabled on the extension (see “Is Agent” on page 40).
3. Configure the agent’s log-in settings (see “Options” on page 70).
4. On the navigation pane, click **Call Queues**. The Call Queues page is displayed and existing Queues are listed on the left.



Queue Extension Configuration

Queues:

*98 – Check Voicemail
401 – FXS 1
402 – FXS 2
403 – FXS 3
404 – FXS 4
405 – FXS 5
406 – FXS 6
407 – OUT 7
408 – OUT 8
409 – IN 9
410 – IN 10
411 – IN 11
412 – IN 12

Queue:

Full Name:

Strategy : ringall

Agents:

Queue Options:

TimeOut:

Wrapup Time:

Max Len:

Music On Hold: default

□ Auto Fill:

□ Auto Pause:

□ JoinEmpty:

□ LeaveWhenEmpty:

□ Report Hold Time:

5. Either:
   - click New to configure a new Call Queue
   - select an existing Call Queue to edit the configuration
6. In the **Queue field**, enter the Queue’s extension number.
7. In the **Full Name** field, enter a descriptive name.
8. From the **Strategy** dropdown menu, select a call routing option:

   Ring All
   When a new call is received, this option rings every agent who is not on an active call.

   Round Robin
   This option allocates new calls to agents in turn.

Least Recent
This option allocates an incoming call to the agent who has been without a call the longest.

Fewest Calls
This option allocates an incoming call to the agent who has handled the fewest calls.

Random
This option allocates calls randomly.

RrMemory
This option is similar to Round Robin except that it records which agent received the last call and, following a period during which calls were not received, resumes with the next agent in sequence.

9. In the **Agents** field, select the Users that assigned to the queue.
10. Configure the following **Queue Options**:

Timeout
Enter the length of time an agent's phone will ring before the call is forwarded to another agent.

Wrapup Time
This setting defines a length of time which—following an agent's completing one call—they remain unavailable to receive another call.

Max Len
Enter the maximum number of callers allowed in the queue before they are either sent to voice-mail or receive a busy signal.

Music on Hold
From the dropdown menu, select the music that is played while a call is on hold.

Auto Fill
When selected, multiple calls arriving simultaneously are immediately forwarded to agents.

Auto Pause
When selected, if an agent fails to answer a call, subsequent calls are temporarily not sent to the agent.

Join Empty
When selected, callers can enter a queue even if no agents are logged into it. When cleared, callers cannot enter a queue until at least one agent is present.

Leave When Empty
When selected, callers cannot enter a queue if there are no available agents.

Report Hold Time
When selected, an announcement tells the agent how long the call was waiting before it was received by the agent.

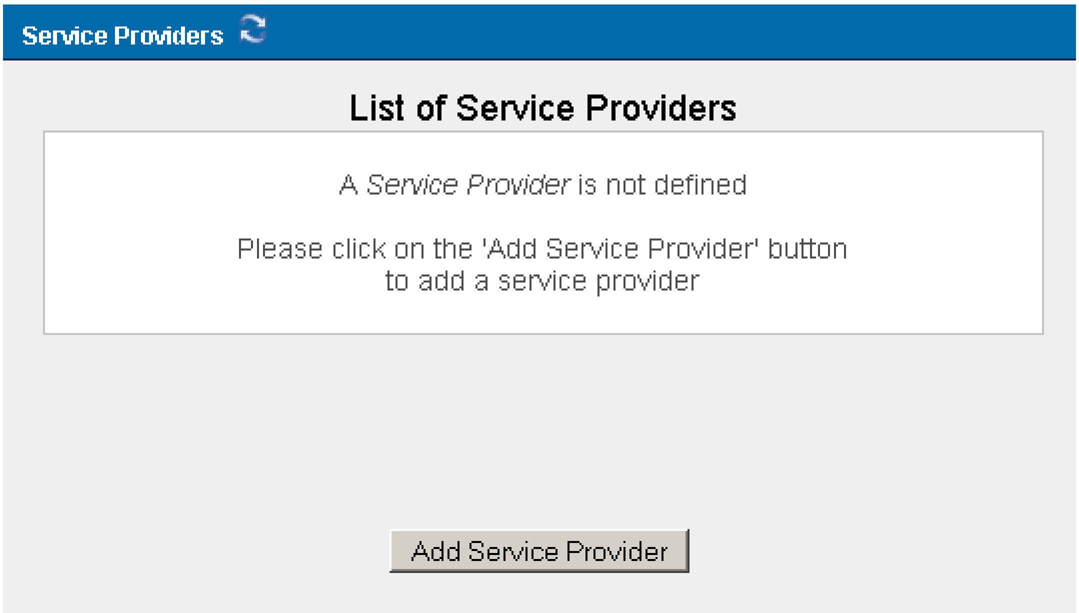
11. Optionally, test the changes.
12. To permanently save the changes, return to the System Configuration browser-window and click **Save Configuration**.

Adding Service Providers

The PBX communicates with the external world via Service Providers. Use the Service Providers screen to define the Service Providers.

Each service provider is actually either an analog (FXO) or a VoIP (SIP or IAX2) trunk.

1. Run the PBX Configurator (see "Running the PBX Configurator" on page 34).
2. On the navigation pane, click **Service Providers**. A list of service providers is displayed.



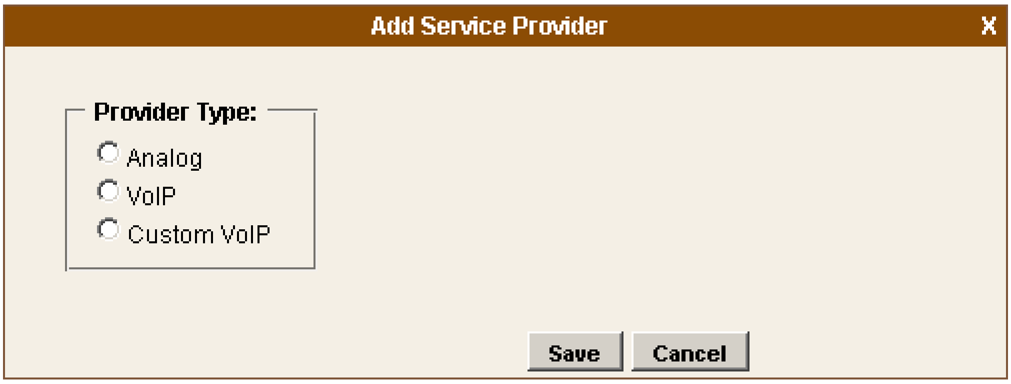
Service Providers

List of Service Providers

A *Service Provider* is not defined

Please click on the 'Add Service Provider' button to add a service provider

3. Click **Add Service Provider**. An Add Service Provider dialog box is displayed. The parameters displayed in this dialog box vary according to provider type.



Add Service Provider

Provider Type:

○ Analog
○ VoIP
○ Custom VoIP

4. Select the **Provider Type**. The associated Add Service Provider dialog box is displayed.

Analog

Select Analog to define the analog ports that have access to this service provider. The dialog box displays the available analog ports. Select one or more analog ports.

VoIP

Select a VoIP service provider from a list of pre-configured providers and enter your user name and password.

Custom VoIP

Select Custom VoIP to customize a VoIP definition. Then, enter the following:

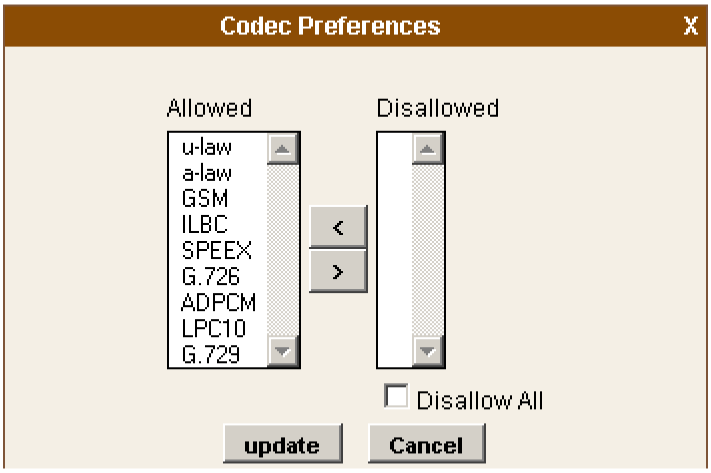
- Comment
  Enter a distinct and identifiable comment, such as the name of the custom VoIP definition.
- Protocol
  Specify either the IAX or the SIP protocol.
- Register
  Select Register to register the IP address with the service provider.
- Host
  Enter the IP address or DNS host name of the service provider.
- Username
  Enter the user name associated with your provider account.
- Password
  Enter the password associated with your provider account.

5. Click **Save**. The new service provider appears on the list of providers on the Service Providers page.
6. If necessary, from the **Options** dropdown menu, refine the configuration. Some options are available only for specific types of service providers.

Codecs

All codecs are allowed by default. To disable a codec:

a. From the Options dropdown menu, select **Codecs**. The Codec Preferences screen is displayed.



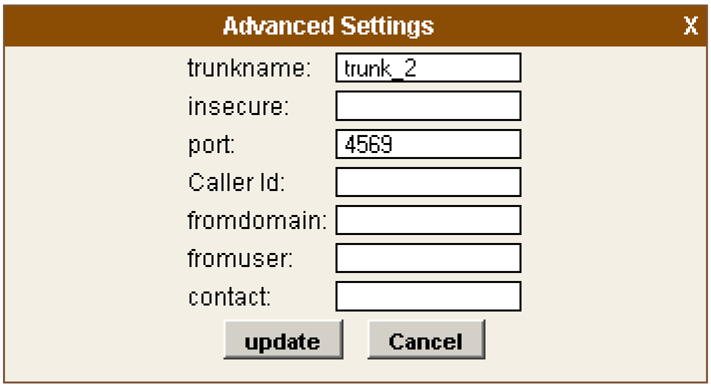
b. Select a codec.

c. Use the arrow buttons to move the codec to the opposite list.

d. Click **Update**.

Advanced

a. From the Options dropdown menu, select **Advanced**.



trunkname: trunk_2
insecure:
port: 4569
Caller Id:
fromdomain:
fromuser:
contact:

b. Configure the following advanced parameters:

- Trunkname
  Enter a distinct name for the trunk.
- Insecure
  Define the security level of the connection to the service provider. Options are: very, yes, no, invite, and port. You can define both port and invite by separating them with a comma, for example: port,invite.
  - Port
    Allows authentication of the incoming peer by IP address only. Select this option if you want to use several accounts from the same provider.
  - Invite
    Does not require authentication of incoming INVITEs. Define this option to receive incoming calls from the VoIP provider.
- Port
  Enter the port through which the register request is sent to the service provider. The default port is 5060.
- Caller ID
  Enter the Caller ID.
- Fromdomain
  Enter the domain name you want to appear in the "From:" header field of SIP messages that are sent by the PBX to providers.
- Fromuser
  Enter a SIP user name you want to appear in the "From:" header field of SIP messages that are sent by the PBX to provider.

- Contact
  Enter a SIP user name you want to appear in the “Contact:” header field of SIP messages that are sent by the PBX to provider.

c. Click **Update**.

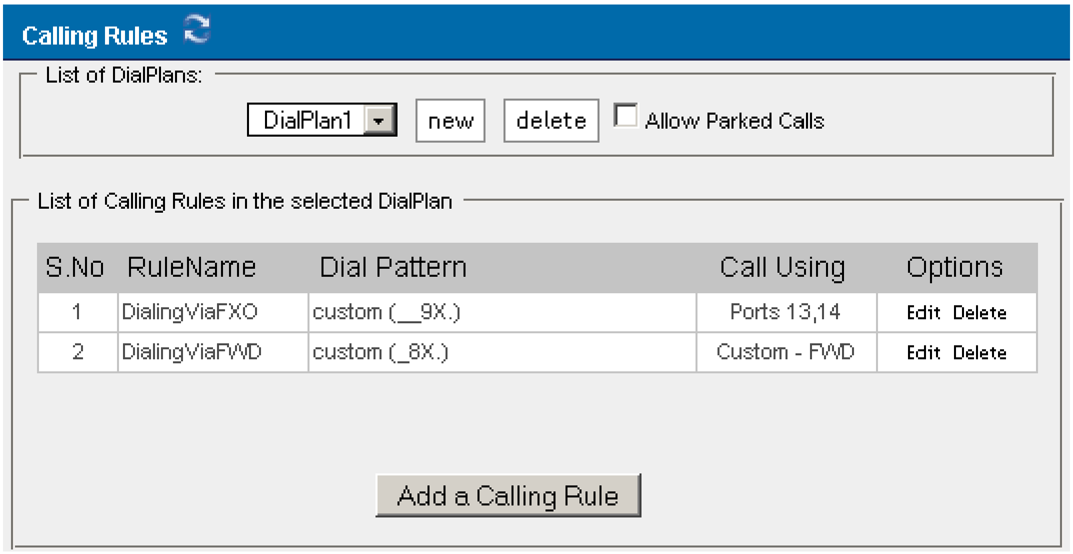
7. Click **Activate Changes**. The changes are immediately available.
8. Optionally, test the changes.
9. To permanently save the changes, return to the System Configuration browser-window and click **Save Configuration**.

Calling Rules

Use Calling Rules to route outbound calls to specific service providers. To add a Calling Rule:

1. Run the PBX Configurator (see “Running the PBX Configurator” on page 34).
2. On the navigation pane, click **Calling Rules**. A list of existing calling rules for a specific Dial Plan is displayed.

Note: *A Dial Plan is a collection of Calling Rules.*



Calling Rules

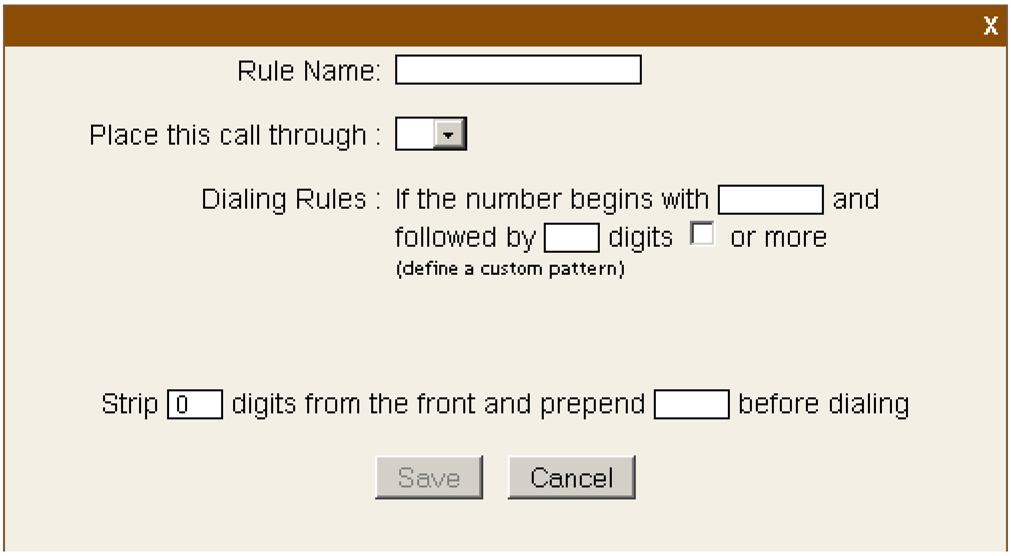
List of DialPlans:

DialPlan1 ▾ | new | delete | □ Allow Parked Calls

List of Calling Rules in the selected DialPlan

S.No	RuleName	Dial Pattern	Call Using	Options
1	DialingViaFXO	custom (__9X.)	Ports 13,14	Edit Delete
2	DialingViaFWD	custom (_8X.)	Custom - FWD	Edit Delete

3. From the List of Dial Plans box, either select an existing Dial Plan from the dropdown menu or click New to create a new Dial Plan.
4. Click **Add a Calling Rule**. The Add a Calling Rule dialog box is displayed.



Rule Name:

Place this call through : ▾

Dialing Rules : If the number begins with ____ and followed by ____ digits □ or more
(define a custom pattern)

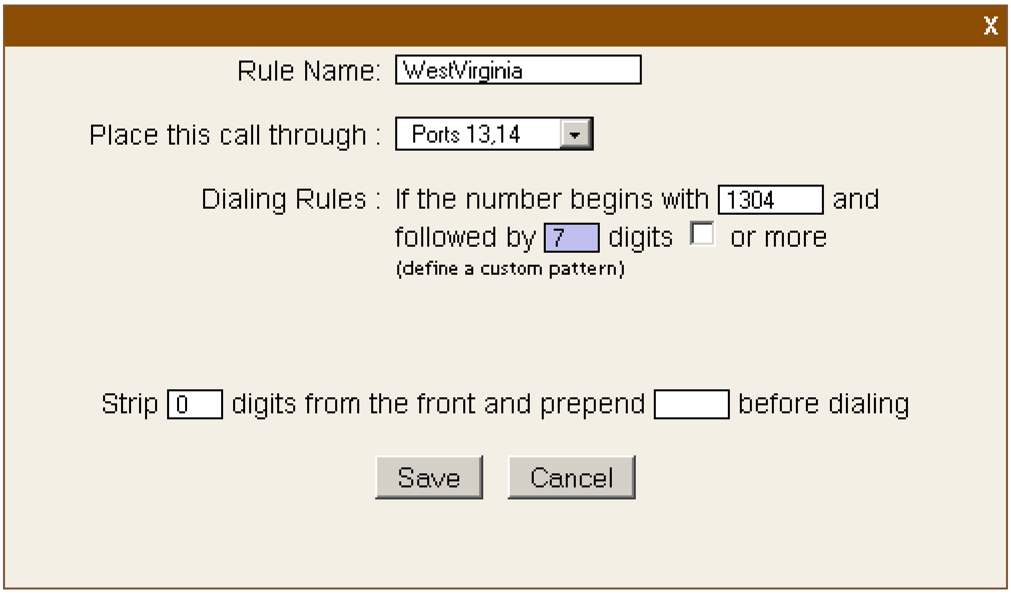
Strip 0 digits from the front and prepend ____ before dialing

5. Define the following parameters:

Rule Name
Enter a descriptive name.

Place this Call Through
Select the service provider to which the rule applies.

Dialing Rules
This sentence defines the phone number that the rule is applied to. Enter numbers in the fields. In the following example, calls to West Virginia are affected:



Rule Name: WestVirginia

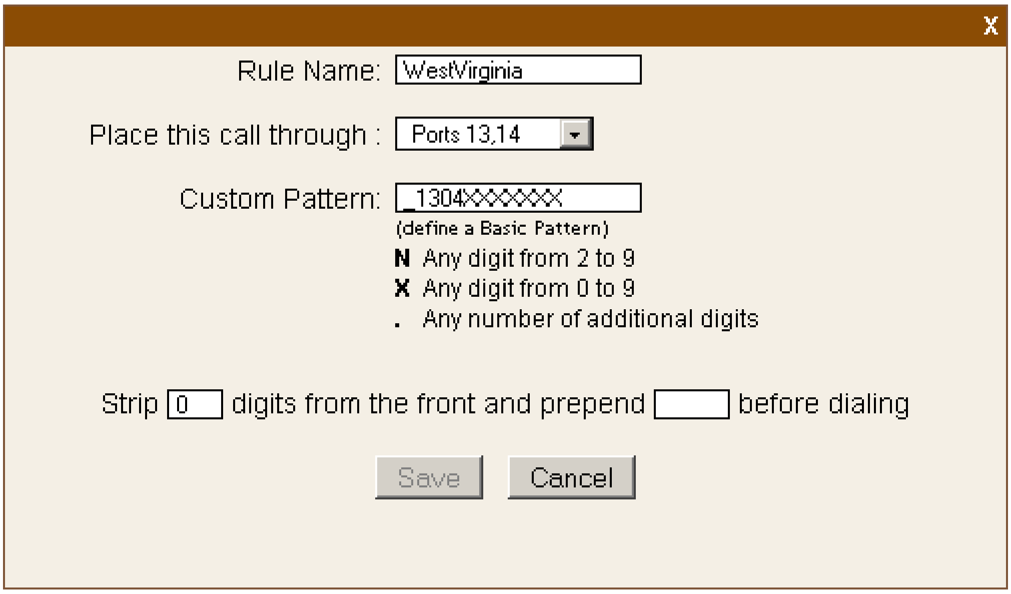
Place this call through : Ports 13,14

Dialing Rules : If the number begins with 1304 and followed by 7 digits □ or more

(define a custom pattern)

Strip 0 digits from the front and prepend before dialing

Alternatively, you can click **define a custom pattern**. The Custom Pattern field is displayed.



Rule Name: WestVirginia

Place this call through : Ports 13,14

Custom Pattern: _1304XXXXXXX

(define a Basic Pattern)

N Any digit from 2 to 9

X Any digit from 0 to 9

. Any number of additional digits

Strip 0 digits from the front and prepend before dialing

In the Custom Pattern field, define your own pattern using the following rules:

- the pattern may consists of digits and special wildcard characters
- if the pattern contains at least one wildcard character then it must have the underscore ('_') character in the beginning
- wildcard character X matches any digit from 0-9
- wildcard character Z matches any digit from 1-9
- wildcard character N matches any digit from 2-9
- wildcard character dot (.) matches one or more any characters

For example:

- Pattern: _9054XXXXXXX
  Matches any phone number that has 9054 at the beginning and then 7 digits.
- Pattern: _9054.
  Matches any phone number that has 9054 at the beginning and the one or more other digits.

Detailed description of the phone number patterns you can find at:
http://www.voip-info.org/wiki/index.php?page=Asterisk+Dialplan+Patterns

⚠ ***WARNING!*** *Incorrect building of the dialing rules may cause call failures while dialing from the analog phones. See "FXS/FXO Channel Related Issues" on page 75 in the Troubleshooting section for further details.*

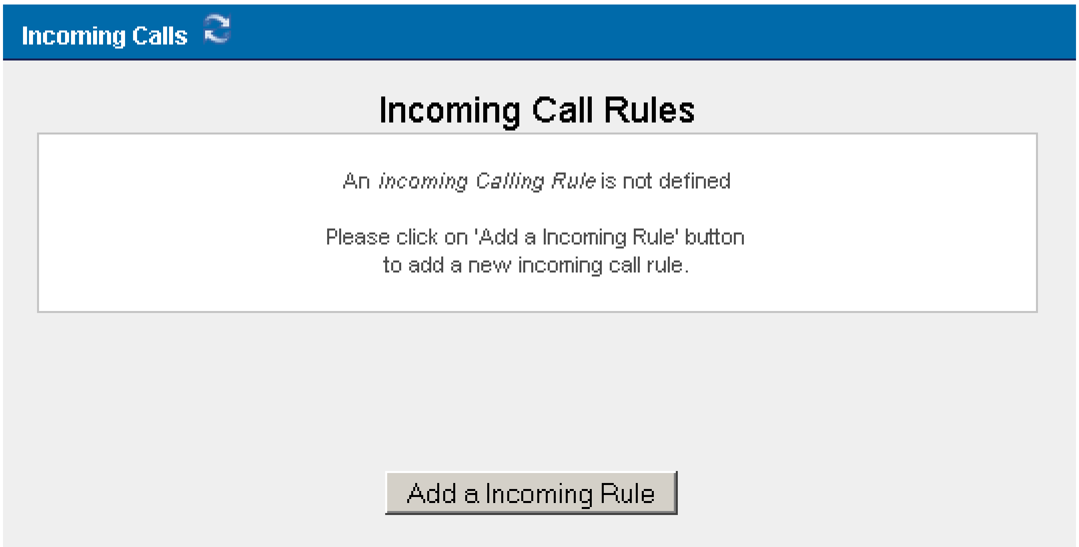
Strip / Prepend
This sentence defines the changes that are made to the original number. Strip defines the quantity of numbers that are deleted from the beginning of the original number. Prepend defines digits that added to the beginning of the number after the strip is complete.

6. Click **Save**.
7. Click **Activate Changes**. The changes are immediately available.
8. Optionally, test the changes.
9. To permanently save the changes, return to the System Configuration browser-window and click **Save Configuration**.

Incoming Calls

Use Incoming Calls to create rules that route incoming calls to a specific extension. To create an Incoming Call rule:

1. Run the PBX Configurator (see “Running the PBX Configurator” on page 34).
2. On the navigation pane, click **Incoming Calls**. A list of existing incoming call rules is displayed.



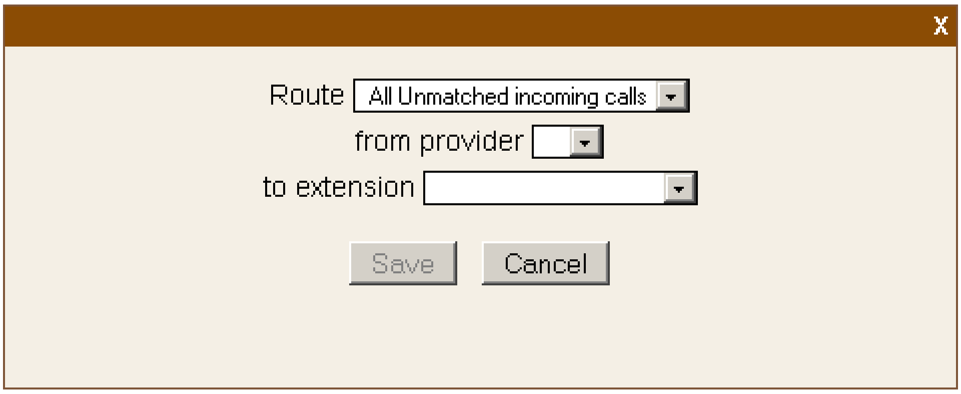
Incoming Calls

Incoming Call Rules

An *incoming Calling Rule* is not defined

Please click on 'Add a Incoming Rule' button to add a new incoming call rule.

3. Click **Add an Incoming Rule**. The Add an Incoming Rule dialog box is displayed.



Route: All Unmatched incoming calls

from provider:

to extension:

4. Define the following parameters:

   Route
   From the dropdown menu, select either All Unmatched Calls or Calls Which Match.

   From Provider
   Select a provider.

   To Extension
   Enter the extension to which the calls will be routed.

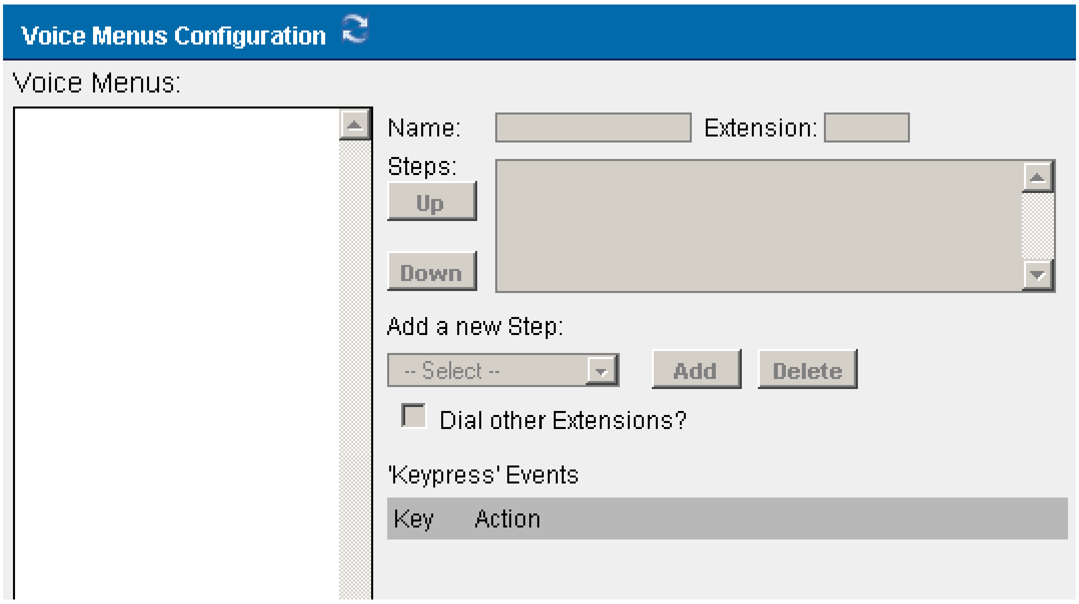
5. Click **Save**.
6. Click **Activate Changes**. The changes are immediately available.
7. Optionally, test the changes.
8. To permanently save the changes, return to the System Configuration browser-window and click **Save Configuration**.

Voice Menus

Use the Voice Menus page to create and edit Interactive Voice Responses (IVR) or voice menus.

To create a new IVR:

1. Run the PBX Configurator (see “Running the PBX Configurator” on page 34).
2. On the navigation pane, click **Voice Menus**. The Voice Menu Configuration screen is displayed.



3. Click **New**.
4. Enter a descriptive name.
5. Optionally, enter the Extension that plays the Voice Menu.
6. Click **Save**.
7. From the **Add a New Step** dropdown menu, select an action. The options are:

   Answer
   This step is added automatically and answers the incoming call.

   Authenticate
   This step is used to restrict access to one or more areas of the system. Use of a PIN code is associated with this step.

   Background
   This step is used to play an audio file in the background while waiting for the caller to enter an extension or number. Playback ends once the user begins to enter a number. Scroll through the list of pre-recorded sound files and select a file.

Record your own message using the "Record a Menu" screen (see "Record a Menu" on page 66). For example: "Hello. Thank you for calling ABC company. For technical support dial 1, for R&D dial 2."

Busytone
Select the Busytone option if there is a step in the process at which a busy signal should be played.

Congestion
Select the Congestion option if there is a step in the process at which a Congestion signal should be played.

Digit Timeout
Use the Digit Timeout option to set the maximum amount of time allowed between key presses. If a full extension is not entered in the specified time, the entry will be considered invalid.

DISA
Use the DISA option to allow callers from outside the system access to an internal dial tone and thereby place calls from within the system. A passcode is required.

Response Timeout
If a caller does not enter a response with the time specified in this field, the call will terminate.

Playback
This option plays a selected sound file. However, this option does not listen for a KeyPress event, and proceeds to the next step in the list.

Wait
This option pauses the execution of steps in the voice menu list for the number of seconds specified.

WaitExten
This option gives a caller a specified amount of time to enter an extension.

Goto Menu
This option sends a caller to a specified voice menu.

Goto Directory
This option sends a caller to the system phone directory. This gives the user the chance to select a user name from the directory.

Goto Extension
This option sends a caller to a specified extension. Select the extension from the available list.

Dial RingGroup
This option dials a specified RingGroup.

Hangup
This option ends the call.

8. Select or clear the **Dial Other Extensions** checkbox. When selected, the user can exit the menu selections and dial an extension within the system.
9. Define the Keypress Event actions for digits 0–9, *, #, t, and i.

t = time out	This is an event that occurs when the user did not dial a digit during a timeout that may be defined, for example, in the WaitExten action step.
i = invalid	This is an event that occurs when the user dialed a keypress event digit that has not been configured.

*Note: We recommend **not** using the t key and i key for specific actions. The action associated with the t key should be the action if a user response has timed-out. The action associated with the i key should be the action if a user makes an invalid entry.*

The options are:

Disabled
The key is disabled.

Goto Menu
Pressing a key assigned this option transfers the caller to a specific menu.

Goto Extension
Pressing a key assigned this option transfers the caller to a specific extension. For example, you can define that if the caller pressed 1 then the PBX must connect the caller to extension 1734.

Custom
Use this option to define calling to an Asterisk application or an AGI script. A strong knowledge of Asterisk configuration options is required to use this option.

Hangup
Pressing a key assigned this option ends the call.

Play Invalid
Pressing a key assigned this option tells the caller that they have made an invalid entry.

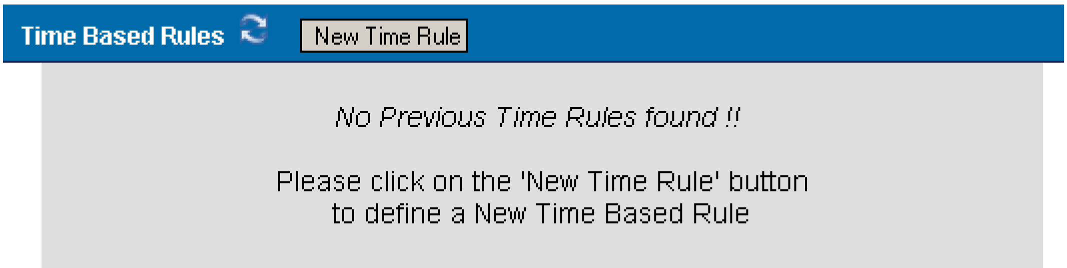
10. Click **Save**.
11. Click **Activate Changes**. The changes are immediately available.
12. Optionally, test the changes.
13. To permanently save the changes, return to the System Configuration browser-window and click **Save Configuration**.

Time Based Rules

Use Time Based Rules to define the route of incoming calls. Rules can be created based on the calendar and the time of day.

To create a new Time Based Rule:

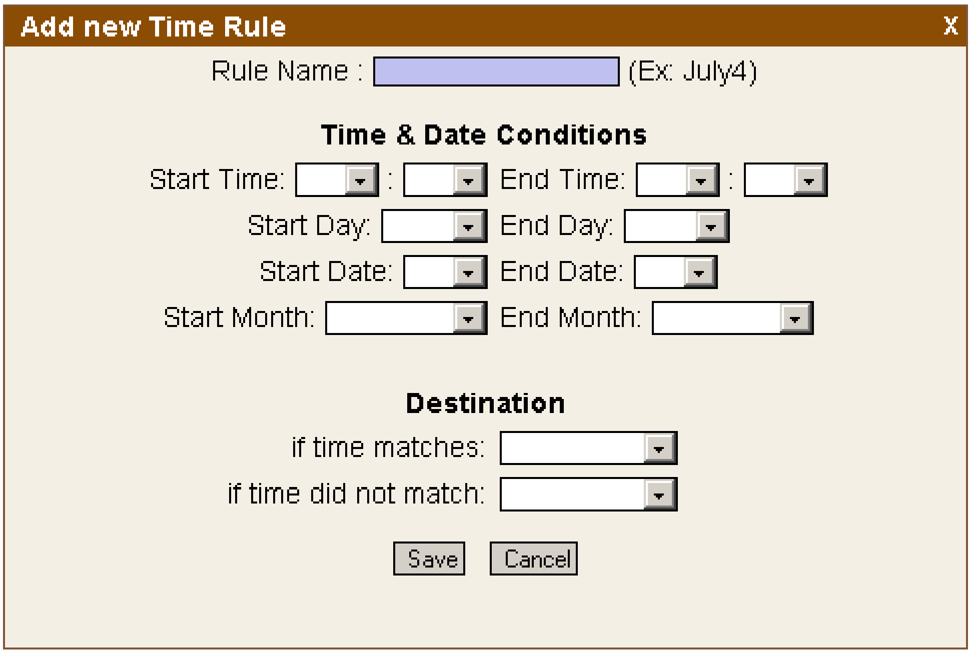
1. Run the PBX Configurator (see “Running the PBX Configurator” on page 34).
2. On the navigation pane, click **Time Based Rule**. The Time Based Rules screen is displayed.



No Previous Time Rules found !!

Please click on the 'New Time Rule' button
to define a New Time Based Rule

3. Click **New Time Rule**. The Add New Time Rule dialog box is displayed.



4. Enter a descriptive name for the new rule.
5. Enter the Time and Date Conditions:

   Start Time / End Time
   The rule is applied beginning from the hour selected from Start Time dropdown menus. The rule is canceled at the hour selected from the End Time dropdown menus.

Start Day / End Day
The rule is applied beginning from the day of the week selected from Start Day dropdown menu. The rule is canceled on the day of the week selected from the End Day dropdown menu.

Start Date / End Date
The rule is applied beginning from the date of the month selected from Start Date dropdown menu. The rule is canceled at the date of the month selected from the End Date dropdown menu.

Start Month / End Month
The rule is applied beginning from the month selected from Start Month dropdown menu. The rule is canceled at the hour selected from the End Month dropdown menu.

6. Using the dropdown menus, enter the Destination parameters:

   If time matches

   If time did not match

7. Click **Save**. The Add New Time Rule dialog box closes and focus returns to the Time Based Rules screen.
8. Click **Activate Changes**. The changes are immediately available.
9. Optionally, test the changes.
10. To permanently save the changes, return to the System Configuration browser-window and click **Save Configuration**.

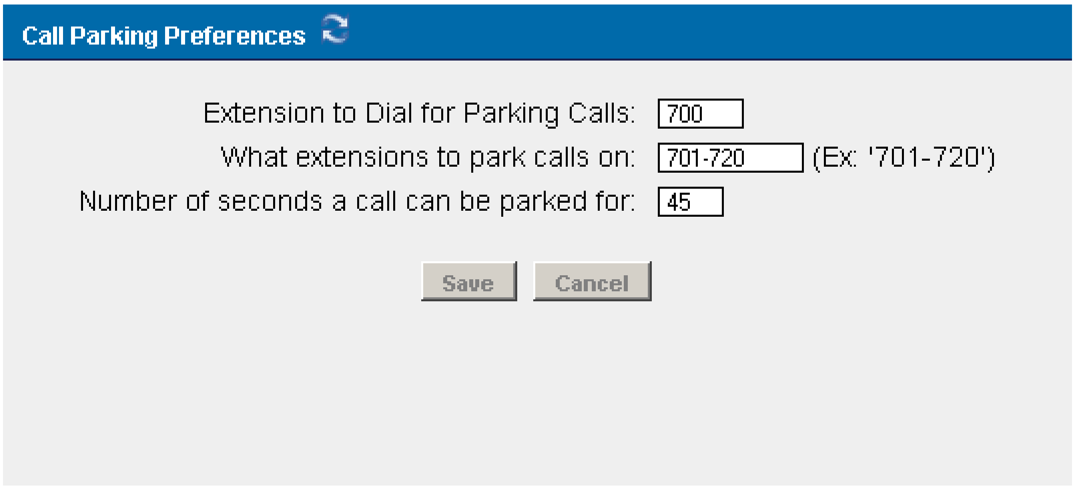
Call Parking

Call Parking places a call on hold such that it can be taken off hold from another extension.

Defining Call Parking Extensions

To define the Call Parking parameters:

1. Run the PBX Configurator (see “Running the PBX Configurator” on page 34).
2. On the navigation pane, click **Call Parking**. The Call Parking screen is displayed.



Call Parking Preferences

Extension to Dial for Parking Calls: 700

What extensions to park calls on: 701-720 (Ex: '701-720')

Number of seconds a call can be parked for: 45

3. In the **Extension to Dial for Parking Calls** field, enter the number dialed to put a call on hold.
4. In the **What Extensions to Park Calls On** field, enter the extensions that will hold the calls. These are the extensions that the next user will call in order to retrieve the call.
5. In the **Number of Seconds a Call Can Be Parked** field, enter the number of seconds a call can remain on hold. After this length of time has elapsed, the call will ring the originating extension.
6. Click **Save**.
7. Click **Activate Changes**. The changes are immediately available.
8. Optionally, test the changes.
9. To permanently save the changes, return to the System Configuration browser-window and click **Save Configuration**.

Using Call Parking

Call Parking can be executed using either an analog or VoIP phone.

- Analog phone
  Press the flash button, or quickly click the hook switch. Then, wait for a dial tone and dial the extension.
- VoIP Phone
  The method using a VoIP phone will vary depending on the phone. In general, initiate the transfer, dial the call parking extension, then complete the transfer (such as by click send).

In both cases, the XR1000 will announce a number. This is the number that must be entered to retrieve the call.

To retrieve the call, pickup a phone and dial the parking number.

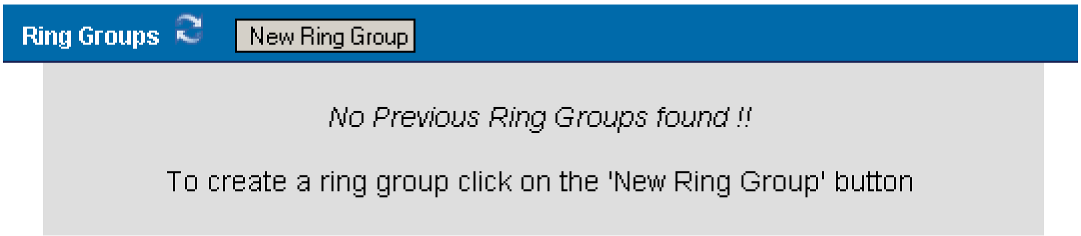
Note: *To park a call, use attended transfer functions. Using a blind transfer function will not provide the parking number to the person parking the call.*

Ring Groups

Use Ring Groups to simultaneously (or in a ring order) ring a group of phones.

To create a new ring group:

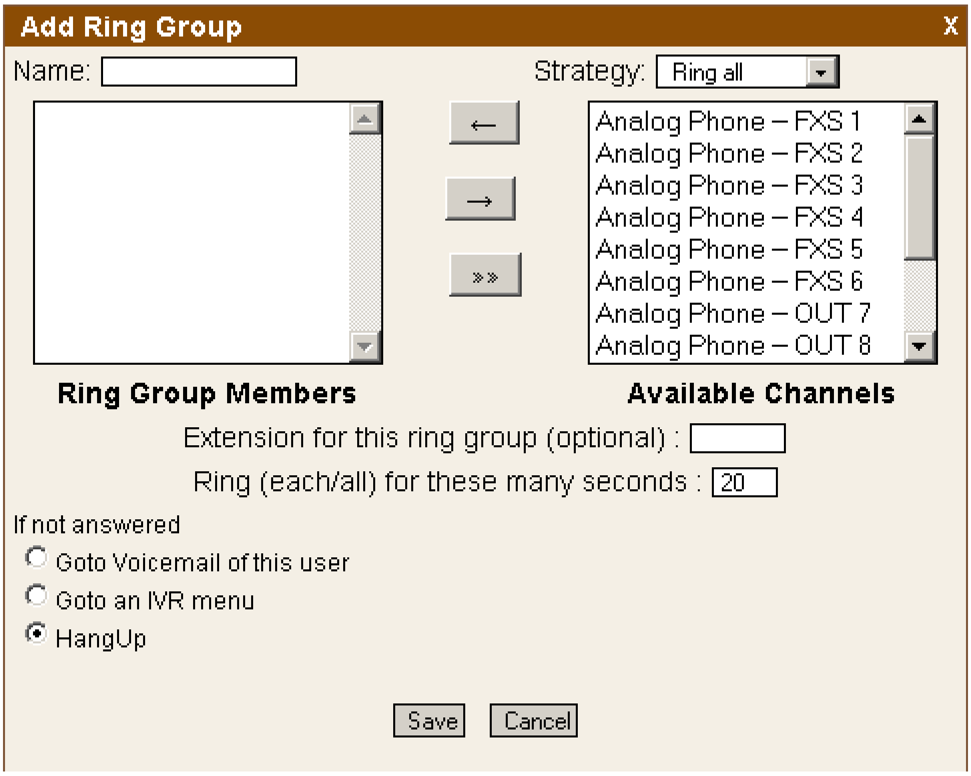
1. Run the PBX Configurator (see “Running the PBX Configurator” on page 34).
2. On the navigation pane, click **Ring Groups**. The Ring Groups screen is displayed.



No Previous Ring Groups found !!

To create a ring group click on the 'New Ring Group' button

3. Click **New Ring Group**. The Add Ring Group dialog box is displayed.



Name:

Strategy: Ring all

Analog Phone – FXS 1
Analog Phone – FXS 2
Analog Phone – FXS 3
Analog Phone – FXS 4
Analog Phone – FXS 5
Analog Phone – FXS 6
Analog Phone – OUT 7
Analog Phone – OUT 8

Extension for this ring group (optional) :

Ring (each/all) for these many seconds : 20

If not answered

○ Goto Voicemail of this user
○ Goto an IVR menu
● HangUp

4. Enter a distinctive name for the group.
5. From the **Strategy** dropdown menu, select either:

   Ring All
   All phones in the defined group ring at the same time.

   Ring Order
   Phones ring in a sequence determined by the order of the users or trunks in the group.

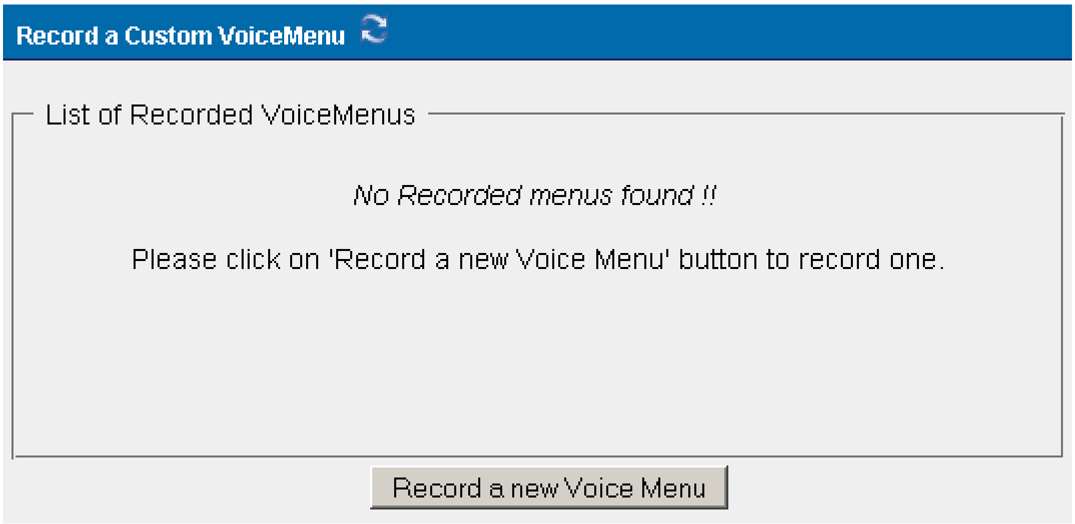
6. Select the members of the ring group:
   a. From the Available Channels list, click on a user extension or trunk.
   b. Click the left arrow. The extension is moved to Ring Group Members list.
   c. Repeat the process for additional channels.
7. Optionally, in the **Extension for this Ring Group** field, enter an extension that can be dialed to ring all members of the group.
8. Enter the number of seconds that each phone should ring.
9. Select an action that occurs if no one answers the call.
10. Click **Save**.
11. Click **Activate Changes**. The changes are immediately available.
12. Optionally, test the changes.
13. To permanently save the changes, return to the System Configuration browser-window and click **Save Configuration**.

Record a Menu

Use the Record a Menu screen to record a custom menu prompts to be used in a Voice Menu. Recording are made using a telephone.

To record a new voice menu option:

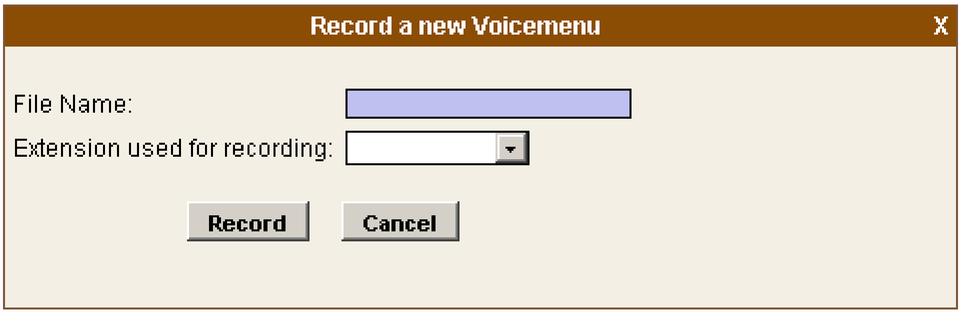
1. Run the PBX Configurator (see “Running the PBX Configurator” on page 34).
2. On the navigation pane, click **Record a Menu**. The Record a Menu screen is displayed.



No Recorded menus found !!

Please click on 'Record a new Voice Menu' button to record one.

3. Click **New**. The Record a New Voice Menu dialog box is displayed.



4. Enter a distinctive name for the new voice menu.
5. From the **Extension Used for Recording** dropdown menu, select the extension the user will use for recording the prompt. This is the phone-extension referred to in step number 7.
6. Click **Record**. The PBX will ring to the defined extension and prompt you to start recording of your messages.
7. Pick up the phone and record the messages.
8. Hang-up the phone.

The new recorded messages are available from the Voice Menu (see “Voice Menus” on page 57).

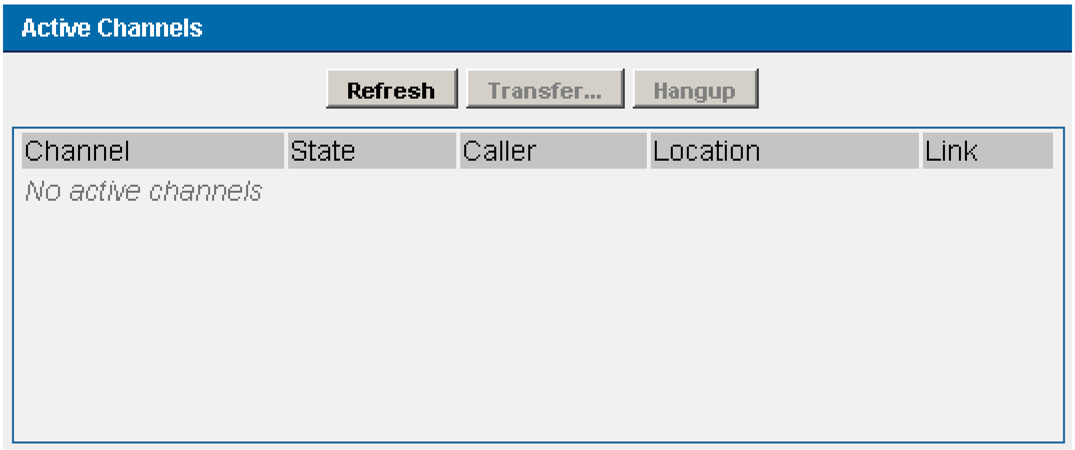
Administrator Monitors and Tools

The XR1000 PBX Configurator includes a set of monitors and tools.

Active Channels

The Active Channels monitor displays a remote view of the active calls and devices, the activity of the server, and can be refreshed to view the progression of calls.

To review the activity on the XR1000, click **Active Channels**. The Active Channels screen is displayed.



Channel	State	Caller	Location	Link
No active channels				

Asterisk Logs

To review the log entries that have been saved in /var/log/asterisk/messages file, on the navigation pane, click **Asterisk Logs**. By default, the XR1000 ***does not*** add entries to this file.

To enable it:

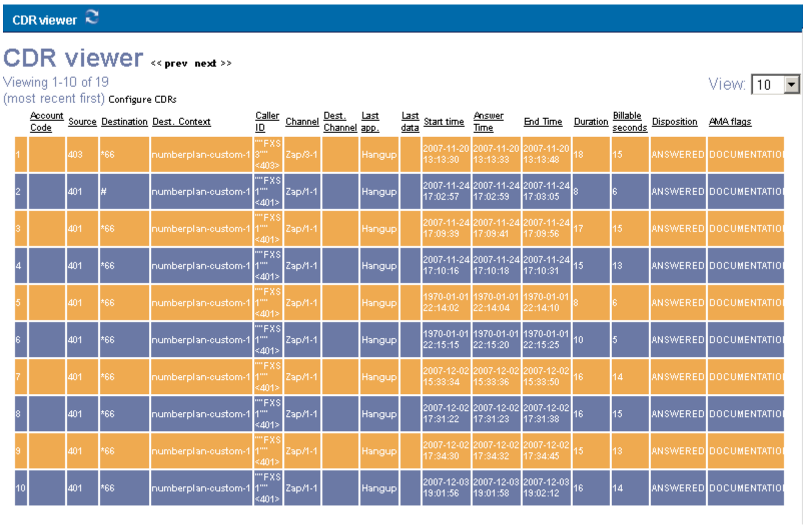
1. On the navigation pane, click **File Editor**. The File Editor is displayed.
2. Access the file named `logger.conf` (see “File Editor” on page 69).
3. Add the line `messages` to the section `[logfiles]`.

   For example, if you want Asterisk to send error messages to the messages file, then you can define: `messages=error`
4. To apply the changes, on the Command Line Interface screen, enter the command `logger reload`.
5. To permanently save the changes, return to the System Configuration browser-window and click **Save Configuration**.

CDR Reader

Use the CDR (Call Detail Records) Reader to review information about current and older calls that were placed through the XR1000.

To open the CDR Reader, on the navigation pane, click **CDR Reader**.



CDR viewer

Viewing 1-10 of 19
(most recent first)

	Account Code	Source	Destination	Dest. Context	Caller ID	Channel	Dest. Channel	Last app.	Last data	Start time	Answer Time	End Time	Duration	Billable seconds	Disposition	AMA flags
1		403	*66	numberplan-custom-1	""FXS 3"" <403>	Zap/3-1		Hangup		2007-11-20 13:13:30	2007-11-20 13:13:33	2007-11-20 13:13:48	18	15	ANSWERED	DOCUMENTATIO
2		401	#	numberplan-custom-1	""FXS 1"" <401>	Zap/1-1		Hangup		2007-11-24 17:02:57	2007-11-24 17:02:59	2007-11-24 17:03:05	8	6	ANSWERED	DOCUMENTATIO
3		401	*66	numberplan-custom-1	""FXS 1"" <401>	Zap/1-1		Hangup		2007-11-24 17:09:39	2007-11-24 17:09:41	2007-11-24 17:09:56	17	15	ANSWERED	DOCUMENTATIO
4		401	*66	numberplan-custom-1	""FXS 1"" <401>	Zap/1-1		Hangup		2007-11-24 17:10:16	2007-11-24 17:10:18	2007-11-24 17:10:31	15	13	ANSWERED	DOCUMENTATIO
5		401	*66	numberplan-custom-1	""FXS 1"" <401>	Zap/1-1		Hangup		1970-01-01 22:14:02	1970-01-01 22:14:04	1970-01-01 22:14:10	8	6	ANSWERED	DOCUMENTATIO
6		401	*66	numberplan-custom-1	""FXS 1"" <401>	Zap/1-1		Hangup		1970-01-01 22:15:15	1970-01-01 22:15:20	1970-01-01 22:15:25	10	5	ANSWERED	DOCUMENTATIO
7		401	*66	numberplan-custom-1	""FXS 1"" <401>	Zap/1-1		Hangup		2007-12-02 15:33:34	2007-12-02 15:33:36	2007-12-02 15:33:50	16	14	ANSWERED	DOCUMENTATIO
8		401	*66	numberplan-custom-1	""FXS 1"" <401>	Zap/1-1		Hangup		2007-12-02 17:31:22	2007-12-02 17:31:23	2007-12-02 17:31:38	16	15	ANSWERED	DOCUMENTATIO
9		401	*66	numberplan-custom-1	""FXS 1"" <401>	Zap/1-1		Hangup		2007-12-02 17:34:30	2007-12-02 17:34:32	2007-12-02 17:34:45	15	13	ANSWERED	DOCUMENTATIO
10		401	*66	numberplan-custom-1	""FXS 1"" <401>	Zap/1-1		Hangup		2007-12-03 19:01:56	2007-12-03 19:01:58	2007-12-03 19:02:12	16	14	ANSWERED	DOCUMENTATIO

Starting from software version 0.95, collecting of CDRs is ***disabled*** by default. To enable the collection of CDRs:

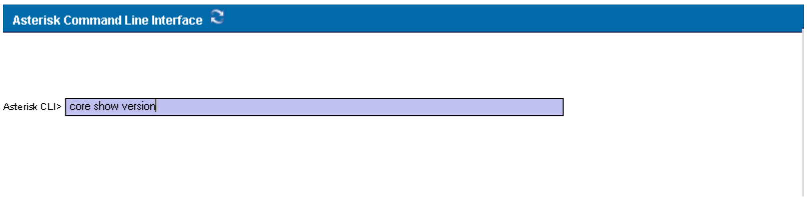
1. On the navigation pane, click **File Editor**. The File Editor is displayed.
2. Access the file named `cdr.conf` (see “File Editor” on page 69).
3. In the general section, define `enable=yes`.
4. Restart the XR1000 to apply the changes.

Command Line Interface

The Asterisk CLI is a command line interface which can be used to issue Asterisk commands.

To open the Asterisk CLI, on the navigation pane, click **Asterisk CLI**.

Type any Asterisk command in the **Asterisk CLI>** field and press Enter. The command output will be displayed.

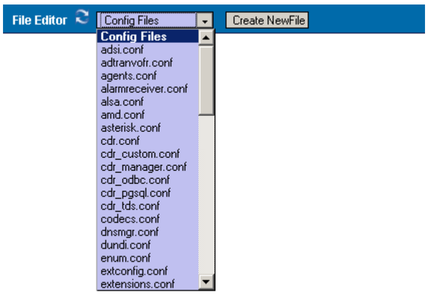


Asterisk CLI> core show version

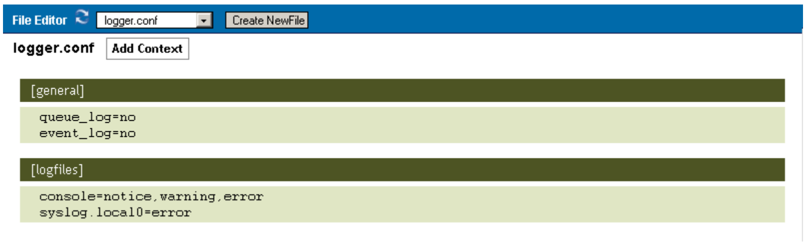
File Editor

Use the File Editor to edit any Asterisk configuration file and to create a new configuration file. Usually, the File Editor is used by advanced Asterisk users.

1. To open the File Editor, on the navigation pane, click **File Editor**. The File Editor page is displayed.
2. Expand the Config File dropdown menu.

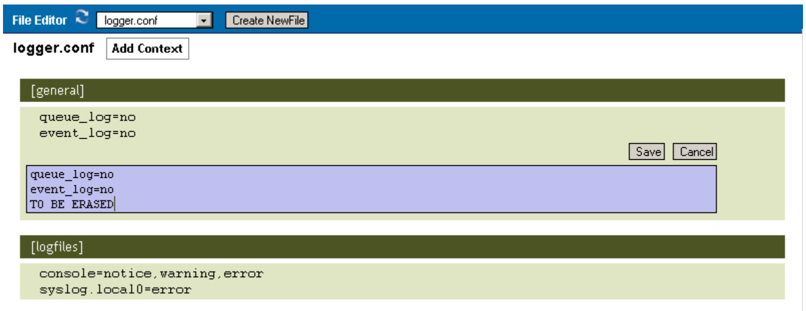


3. Select a file. The file is displayed. This may take a few seconds.



```
[general]
queue_log=no
event_log=no

[logfiles]
console=notice,warning,error
syslog.local0=error
```

4. To edit a section, click within the section. That section's background color turns blue.



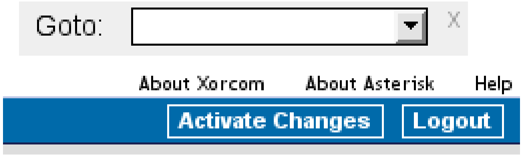
```
[general]
queue_log=no
event_log=no

queue_log=no
event_log=no
TO BE ERASED

[logfiles]
console=notice,warning,error
syslog.local0=error
```

5. Make changes within the blue area and click **Save**.
6. Optionally, to add a new section, click **Add Context**.
7. To permanently save the changes, return to the System Configuration browser-window and click **Save Configuration**.

Options

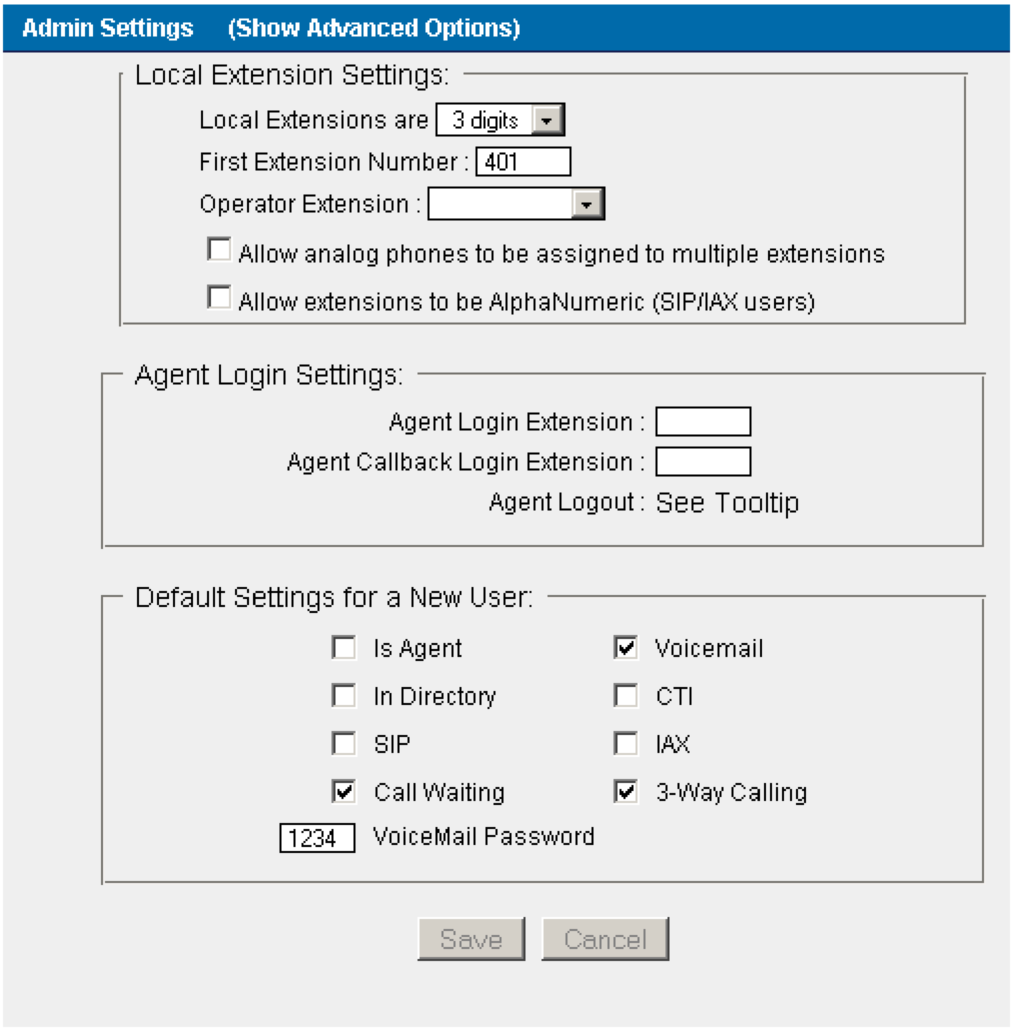
Use the Admin Settings screen to define settings which apply to the entire VoIP network.

Note: The words (Show Advanced Options) appear on the blue title bar. This is a button that, when clicked, opens a dropdown menu entitled Goto: in the upper-right corner. This is displayed in the following picture:



The dropdown menu includes configuration options such as Global SIP Settings, Global IAX settings etc. In most cases, these do not need changed.

1. Run the PBX Configurator (see “Running the PBX Configurator” on page 34).
2. On the navigation pane, click **Options**. The Options screen is displayed.



Local Extension Settings:

Local Extensions are 3 digits

First Extension Number : 401

Operator Extension :

□ Allow analog phones to be assigned to multiple extensions

□ Allow extensions to be AlphaNumeric (SIP/IAX users)

Agent Login Settings:

Agent Login Extension :

Agent Callback Login Extension :

Agent Logout : See Tooltip

Default Settings for a New User:

□ Is Agent ☑ Voicemail

□ In Directory □ CTI

□ SIP □ IAX

☑ Call Waiting ☑ 3-Way Calling

1234 VoiceMail Password

3. Define the Local Extension Settings:

    Local Extensions are
    From the dropdown menu, select the quantity of numbers—or, if AlphaNumeric is enabled, numbers and letters—that a user has to press to connect to an extension.

    First Extension Number
    Extension numbers are sequential. Enter the first number of the sequence.

    Operator Extension
    Enter the extension number pressed to access the network's operator.

    Allow … multiple extensions
    Many analog phones can support more than one extension number. When selected, an analog telephone can be assigned several extension numbers.

    Allow … AlphaNumeric
    When selected, users of SIP or IAX can connect using extensions with letters.

4. Define the Agent Login Settings:

    Agent Login Extension
    This is an extension used by agents to log-in to a specific queue.

    Agent Callback Login Extension
    This is the same as the Agent Login Extension except that you do not have to remain on the line. It is the extension to be dialed for the Agents to log-in to the queues they are apart of.

    Agent Logout
    In case of Callback Login, to log-out, dial the log-in extension and following the voice prompts.

5. Define the Default Settings for a New User:

    Is Agent
    When selected, every new user is also an Agent and may be a member of a Call Queue.

    In Directory
    When selected, new users are automatically added to the directory.

    SIP
    When selected, SIP is the default protocol used for phone calls.

    Call Waiting
    When selected, the Call Waiting feature is enabled for new users.

    Voicemail
    When selected, Voicemail is enabled for new users.

    CTI
    When selected, CTI (Computer Telephone Integration) is enabled for new users. CTI combines data with voice systems thereby enhancing telephone services. For example, automatic number identification (ANI) allows a caller's records to be retrieved from the database while the call is routed to the appropriate party.

IAX

When selected, IAX (Inter-Asterisk Exchange) is the default protocol used for phone calls. IAX is the proprietary Asterisk protocol used to initiate calls and send the data.

3-way Calling

When selected, new users can establish 3-way calls. During a 3-way call, a user can talk with two people at the same time.

VoiceMail Password

Define the password a user enters in order to retrieve their voicemail.

6. Click **Save**.
7. Click **Activate Changes**. The changes are immediately available.
8. Optionally, test the changes.
9. To permanently save the changes, return to the System Configuration browser-window and click **Save Configuration**.

SECTION 5:

SUPPORT AND TROUBLESHOOTING

This section includes:

Rescue, Restore, and Reflashing Firmware

In certain circumstances, you might want to restore the factory default configuration or the default password for the web-based configurator. You will also want to reflash the firmware after a severe system crash, for example, if the device was switched off during the normal firmware update (see “Software Update” on page 31).

⚠ ***WARNING!*** *The Restore button returns the configuration to factory default. Consider backing-up your configuration (see “Backup and Restore” on page 27) before using the Restore button.*

The rear panel of the XR1000 box has a RESCUE button and a RESTORE button. They function as follows:

- Depress the RESTORE button for 5 seconds to reset the configuration to the factory settings.
- Depress the RESCUE button for 5 seconds to reset the password to its default value: 123456.

To reflash the firmware:

1. Download a firmware file onto your computer. The latest firmware version can be downloaded from:
   http://www.xorcom.com/products/xr1000/xr1000_firmware
2. Unplug the unit.
3. Using a tooth-pick, depress the RESCUE button and—while the button is depressed—plug in the unit’s power.
4. After 5 seconds, release the RESCUE button.

   The unit will restart in the rescue mode. IP address 10.0.0.1 is automatically assigned to the WAN interface.
5. Connect to the unit’s WAN interface from another computer. There are several ways of connecting:
   - Connect the device’s WAN RJ-45 connector and your computer to a stand-alone Ethernet hub or switch.
   - If 10.nnn.nnn.nnn IP addresses are not used in your office network, then you can connect the device and computer to the same LAN subnet.
   - Connect your computer by using an Ethernet cross-cable.
6. Set the computer’s IP address to 10.0.0.100, subnet-mask to 255.255.255.0 and default gateway to 10.0.0.1.
7. Start a Firefox browser and enter 10.0.0.1 in the address field. The XR1000 rescue screen will be displayed.
8. Choose the firmware file that was previously stored on your computer and update the XR1000.

   When the update process is finished, the XR1000 will automatically reboot.

Troubleshooting

Please send your suggestions and wish list to info@xorcom.com and we will be glad to contact you and discuss your needs.

For additional support and updated information, visit our web site. To open a support ticket, please use the form available at:

http://www.xorcom.com/static/support-form.html

Our technical support will contact you.

Common PBX Configuration Issues

How can I define the number of times the phone rings before the call is forwarded to the voice mail?

1. Using any standard File Editor, open the file named `extensions.conf`.
2. Go to `context [macro-stdexten]`. The first line in the context looks like the following:

   ```
   exten=s,1,Dial(${ARG2},20)
   ```

   where, 20 is the timeout (in seconds) that defines how long the PBX must try to connect to an FXS or IP extension (SIP/IAX) before it forwards the call to the voice mail service.
3. Change the timeout value.
4. Save and close the file.

FXS/FXO Channel Related Issues

How can I increase the time the Asterisk waits for the next digit when I'm dialing from an FXS extension?

Very frequently, PBX users do not complete dialing the phone number before the PBX starts to make call.

Please be aware of the Asterisk inter-digit timeout handling algorithm for the Zaptel devices:

a. Asterisk waits 16 seconds for the first digit.

b. If there is no ambiguous match, Asterisk waits 8 seconds for the next digit.

c. If there is an ambiguous match, Asterisk waits 3 seconds for the next digit.

Those time-outs are hard-coded and cannot be changed in the Asterisk configuration files.

Following is an example. You have defined two patterns each of which ends with a dot. The first pattern is _9X. — the "x" is followed by a dot. The second pattern is _90044X.

The Asterisk user dials 900-44-208-123-4567.

In this case, the Asterisk will discover the ambiguous match when the user has dialed 900. In other words, the Asterisk will wait for the fourth digit a maximum of three additional seconds.

Therefore, we suggest that you define the dial patterns according to the following two principles:

- The dialing patterns must be as long as possible even when you do not know the exact phone number length. This lowers the probability of a failed dialing attempt.
- Avoid defining default patterns like _9. Patterns like these will overcome your better defined patterns.

How can I enable dialing of an extension number via a system attendant IVR?

You just have to select the **Dial other extensions?** check-box in the Voice Menus Configuration dialog box. See "Voice Menus" on page 57.

While calling out via an FXO line, the call is disconnected after a few ring-back signals due to false busy detection. Can I fix this?

The solution to this is:

1. Using the File Editor (see p. 69), open the file named `zapata.conf`.
2. Define **busycount=8**.
3. Save and close the file.

The PBX users hear an echo when speaking via an FXO port. Can I fix this?

We recommend running FXO Tune. This will calibrate the FXO port electric parameters. See "Tune FXO" on page 28.

It is impossible to initiate call transfer from an FXS that was the “called” call leg.

This can happen when the called party off-hooks the phone immediately after it starts ringing. If the called FXS waits until the second ring to off-hook, then the call transfer will work well.

Therefore, inform users that they should pick up the phone after (or during) the second ring.

Following is a technical description of the problem. When this occurs, the channels status in that situation look like the following:

<table>
<tr><td colspan="4">blackfin*CLI>   show channels</td></tr>
<tr><th>Channel</th><th>Location</th><th>State</th><th>Application (Data)</th></tr>
<tr><td>Zap/1-1</td><td>(None)</td><td>Ringing</td><td>Bridged Call (Zap/3-1)</td></tr>
<tr><td>Zap/3-1</td><td>s@macro-stdexten:1</td><td>Up</td><td>Dial(Zap/1|20)</td></tr>
<tr><td colspan="4">2 active channels
1 active call</td></tr>
</table>

When call transfer works then the channels status look like the following:

<table>
<tr><td colspan="4">blackfin*CLI>   show channels</td></tr>
<tr><th>Channel</th><th>Location</th><th>State</th><th>Application (Data)</th></tr>
<tr><td>Zap/1-1</td><td>(None)</td><td>Up</td><td>Bridged Call (Zap/3-1)</td></tr>
<tr><td>Zap/3-1</td><td>s@macro-stdexten:1</td><td>Up</td><td>Dial(Zap/1|20)</td></tr>
<tr><td colspan="4">2 active channels
1 active call</td></tr>
</table>

We can guess that Asterisk doesn’t switch the called channel to state “Up” when off-hook occurs while the Asterisk transmits the CallerID. The issue requires further investigation.

Symptom

Asterisk rejects any attempt to call any service provider (Zaptel, VoIP) when the call is made from a CISCO SIP phone. At the same time it is possible to call an internal extension.

Solution

Manually add parameter `insecure=port` to the corresponding user section in file `users.conf`.

Technical description

The Asterisk returns SIP response 404 for any INVITE where R-URI's user name is an external telephone number.

The problem occurs because Asterisk tries to find a peer in its internal peer list by comparing both the IP address and the port where the INVITE came from. A peer is added to the list after successful registration. Please note that CISCO phones send REGISTER messages from port 5060 but INVITE messages from other ports. Therefore, the Asterisk can't find the previously registered peer in the list. When parameter `insecure=port` is defined, then Asterisk doesn't check the peer IP port and everything works well.

Symptom

It is impossible to make calls when more than one SIP extension are located behind the same NAT router.

Solution

Manually add parameter `insecure=port` to the corresponding user section in file `users.conf`.

How do I configure the XR1000 to work with Internet and LAN SIP extensions when it is located behind a NAT router with a permanent external IP address.

When the PBX is installed behind a NAT router with a permanent external IP address but should serve SIP endpoints that are located on the Internet, then it is necessary to configure your environment in the following way:

1. Define your PBX as the DMZ host (in terms of D-Link routers) or alternatively, make port forwarding for the following ports:
   - 5060/udp — needed for SIP signaling
   - 10000/udp-20000/udp — needed for voice packets transmit (RTP)
2. In the `sip.conf` file, in the general section, configure the following two PBX global SIP parameters:
   - externip = 62.xxx.xxx.xxx
   - localnet=192.168.0.0/255.255.0.0

   where,

   62.xxx.xxx.xxx = external IP address of the NAT router

192.168.0.0/255.255.0.0 = network address of the LAN side of the NAT router

There are two ways to edit the `sip.conf` file. You can use the File Editor dialog box (see “File Editor” on page 69) to edit the file manually or you can do it in the Global SIP Settings dialog box as described below:

a. Run the PBX Configurator (see “Running the PBX Configurator” on page 34).

b. On the navigation pane, click **Options**. The Options screen is displayed.

a. Click **Show Advanced Options**. You will see the **Goto** dropdown menu in the right upper corner of the window.

b. From the dropdown menu, select **Global SIP Settings**.

c. Scroll down in the Global SIP Settings dialog box to the section named **NAT Support**.

d. Define the external IP and the local network address (e.g., 192.168.0.0/255.255.0.0):

e. To permanently save the changes, return to the System Configuration browser-window, and click **Save Configuration**.

f. Force the Asterisk to reload the SIP configuration or just restart the PBX.

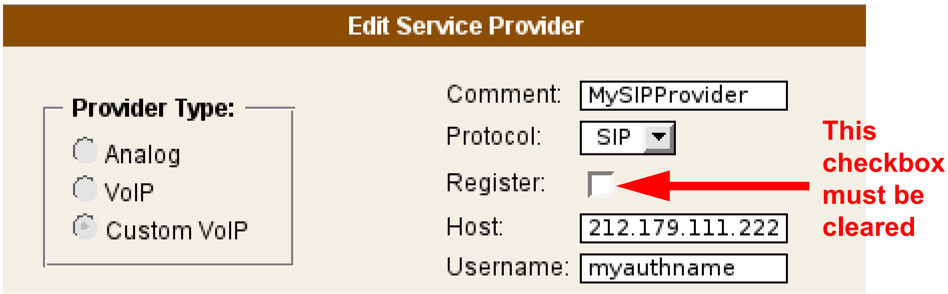
- In order to reload SIP configuration, go to the Asterisk CLI (see “Command Line Interface” on page 68) screen and enter the command `sip reload`.
- Restart Asterisk (see “Restart the Server” on page 32).

After these changes are made, during communication with other SIP endpoints located on the Internet, Asterisk will automatically substitute the router’s external IP address in outgoing SIP messages. During communication with local SIP endpoints, the Asterisk will substitute its local IP address. As result of that, both local and Internet SIP calls can be established.

How can I define a SIP service provider that requires that the user name (e.g., DID number) be different than the SIP authentication name?

Unfortunately, this feature is not supported by the Asterisk GUI and requires both manual changes to some configuration files and service provider configuration.

1. Open the **Add Service Provider** page of the Configurator (see “Adding Service Providers” on page 48).
2. Define or edit the service provider making sure that:
   - the **Register** checkbox is cleared
   - the Username entered is the authentication name that must be used for authentication with SIP server
   - the Password entered is the password that must be used for authentication against SIP server



Edit Service Provider

Provider Type:
○ Analog
○ VoIP
● Custom VoIP

Comment: MySIPProvider
Protocol: SIP
Register: □
Host: 212.179.111.222
Username: myauthname

This checkbox must be cleared

3. On the navigation pane of the PBX configurator, click **Options**. The Options screen is displayed.

4. Click **Show Advanced Options**. You will see the **Goto** dropdown menu in the right upper corner of the window.

5. Define the following settings:

   insecure
   Enter **invite,port**. This directs Asterisk not to perform WWW Authentication for incoming Invite SIP request from this provider. Instead they are authenticated by checking the IP address only.

   Caller Id and **fromuser**
   Enter the SIP user name (DID number).
   Not withstanding the Warning below, if the Caller ID and fromuser parameters are left empty, Asterisk will use the real caller name. For example, if the call was initiated by FXS extension 401, then the outgoing SIP INVITE request will contain 401 as the user name in the FROM header field URI.

WARNING! If Caller ID and fromuser are empty, SIP servers might not be able to authenticate the INVITE request. Many SIP servers expect this field to contain the same user name that was used by the endpoint during the registration process.

6. Open the File Editor ("File Editor" on page 69).

7. Open the `sip.conf` file.

8. Go to the [general] section and define the following line there:

   register=68525:ssssss:myauthname@212.179.111.222/68525

   where,

   68525 = the user name (DID number)

   ssssss = the password

   myauthname = the name to be used for authentication

   212.179.111.222 = the SIP server address

   /68525 = the user name (DID number) that will be used in the CONTACT URL.

9. To permanently save the changes, return to the System Configuration browser-window, and click **Save Configuration**.

Internet SIP extensions cannot register on the XR1000. The LAN sniffing shows that the REGISTER messages arrives to the XR1000 but they do not appear in the Asterisk SIP trace (“sip debug”).

To solve this:

1. Open the File Editor (“File Editor” on page 69).
2. Open the `sip.conf` file.
3. Define `srvlookup=no`.
4. Restart Asterisk.

How can I make a LAN trace on the XR1000?

You can do this using the tcpdump utility that is pre-installed on XR1000:

1. Connect via SSH to theXR1000.
2. Log-in using the User Name **root**. Use the same password that you use to connect to the XR1000 browser-based configurator.
3. Run tcpdump for the appropriate interface and capturing filter. For example, the command below may be used for recording a SIP trace on the XR1000 WAN interface (eth0) and saving it in file /tmp/trc.cap:

   ```
   tcpdump -i eth0 -s 0 -w /tmp/trc.cap 'port 5060'
   ```

4. Press Ctrl-C in order to stop the tcpdump.

Note: *The folder /tmp is mounted on the RAM drive. Therefore, you should be careful of the trace file size.*

5. Copy the file to your computer using any SCP file transfer utility.
6. Review the trace by using the Wireshark (http://www.wireshark.org) analyzer.

I have two accounts at the same SIP provider. How can I handle incoming calls from those accounts? It seems like Asterisk doesn't distinguish between them and routes incoming calls to incorrect destinations.

To solve this, you can create a new context—for example: from-providers—and define the required call routing based on the called number. The new context must be included into the default context. For example:

```
[default]
include => features
include => from-providers
exten => *98,1,VoiceMailMain
...................
[from-providers]
exten = 55791883,1,Goto(default|200|1)
exten = 31015977,1,Goto(default|200|1)
.................
```

PBX Configuration Internals

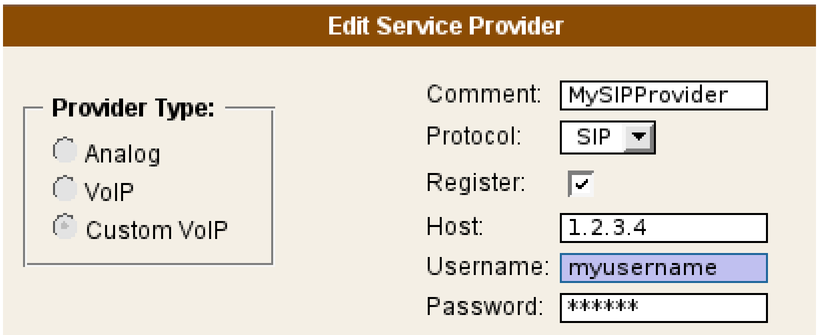
Sometimes you need to make changes in the XR1000 dialing plan by manually editing the Asterisk configuration files. This topic explains how the PBX GUI configurator builds objects in the `users.conf` and `extensions.conf` files.

All of the PBX extensions and service providers are configured in the `users.conf` file. In the example below you can see a definition of extension 401 and two service providers. In the example they are placed side-by-side whereas in the `users.conf` file the second follows the first:

```
[401]
callwaiting = yes
context = numberplan-custom-1
fullname = FXS 1
cid_number = 401
hasagent = no
hasdirectory = no
hasiax = no
hasmanager = no
hassip = no
hasvoicemail = yes
host = dynamic
mailbox = 401
threewaycalling = yes
vmsecret = 1234
secret = 1234
signalling = fxo_ls
zapchan = 1
registeriax = no
registersip = no
canreinvite = no
nat = no
dtmfmode = rfc2833
disallow = all
allow = all


[402]
............
```

```

[trunk_1]
disallow=all
allow = all
context = DID_trunk_1
dialformat = ${EXTEN:1}
hasexten = no
hasiax = no
hassip = yes
host = 1.2.3.4
port = 5060
registeriax = no
registersip = yes
secret = mypassword
trunkname = Custom - MySIP-
Provider
trunkstyle = customvoip
username = myusername

[trunk_2]
callerid = asreceived
context = DID_trunk_2
group = 2
hasexten = no
hasiax = no
hassip = no
trunkname = Ports 13,14
trunkstyle = analog
zapchan = 13,14
```

The provider [trunk_1] is a custom VoIP provider that was defined in the GUI as follows:

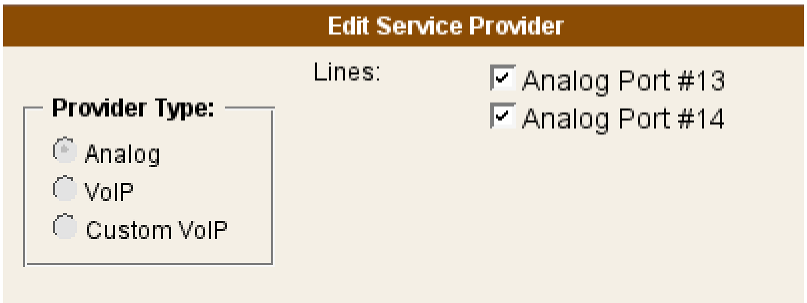


Edit Service Provider

Provider Type:
○ Analog
○ VoIP
● Custom VoIP

Comment: MySIPProvider
Protocol: SIP
Register: ☑
Host: 1.2.3.4
Username: myusername
Password: ******

The provider [trunk_2] is a provider that represents analog (FXO) lines. It was defined as follows:



Edit Service Provider

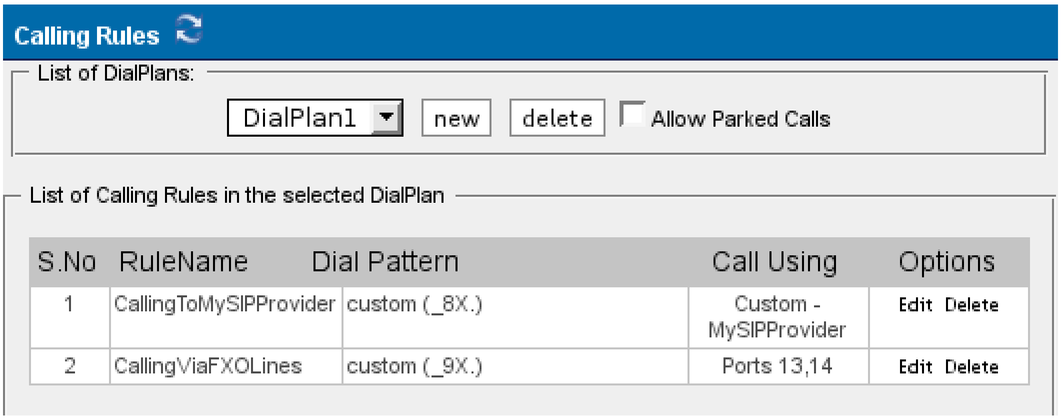
Provider Type:
● Analog
○ VoIP
○ Custom VoIP

Lines: ☑ Analog Port #13
☑ Analog Port #14

Two global variables will be created in the `extensions.conf` file—trunk_1 and trunk_2—and two contexts:

```
[global]
trunk_1 = SIP/trunk_1
trunk_2 = Zap/g2
.......
[DID_trunk_1]
include = default

[DID_trunk_2]
include = default
```

If you create calling rules for ***outbound*** calls via those providers, like the rules in the following picture, then Asterisk creates corresponding rules in the [numberplan-custom-1] context as illustrated below the picture:



Calling Rules

List of DialPlans:

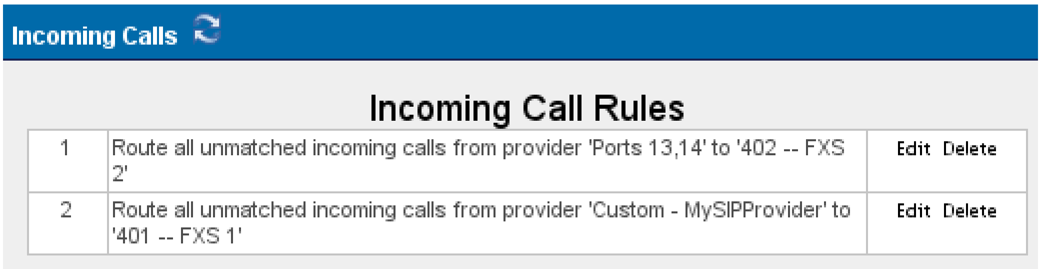
DialPlan1 ▾ | new | delete | □ Allow Parked Calls

List of Calling Rules in the selected DialPlan

S.No	RuleName	Dial Pattern	Call Using	Options
1	CallingToMySIPProvider	custom (_8X.)	Custom - MySIPProvider	Edit Delete
2	CallingViaFXOLines	custom (_9X.)	Ports 13,14	Edit Delete

```
[numberplan-custom-1]
plancomment = DialPlan1
include = default
include = parkedcalls
exten = _8X.,1,Macro(trunkdial,${trunk_1}/${EXTEN:1})
comment = _8X.,1,CallingToMySIPProvider,custom
exten = _9X.,1,Macro(trunkdial,${trunk_2}/${EXTEN:1})
comment = _9X.,1,CallingViaFXOLines,custom
```

If you create calling rules for incoming calls, like the rules in the following picture, then the contexts DID_trunk_1 and DID_trunk_2 will be populated as illustrated below the picture:



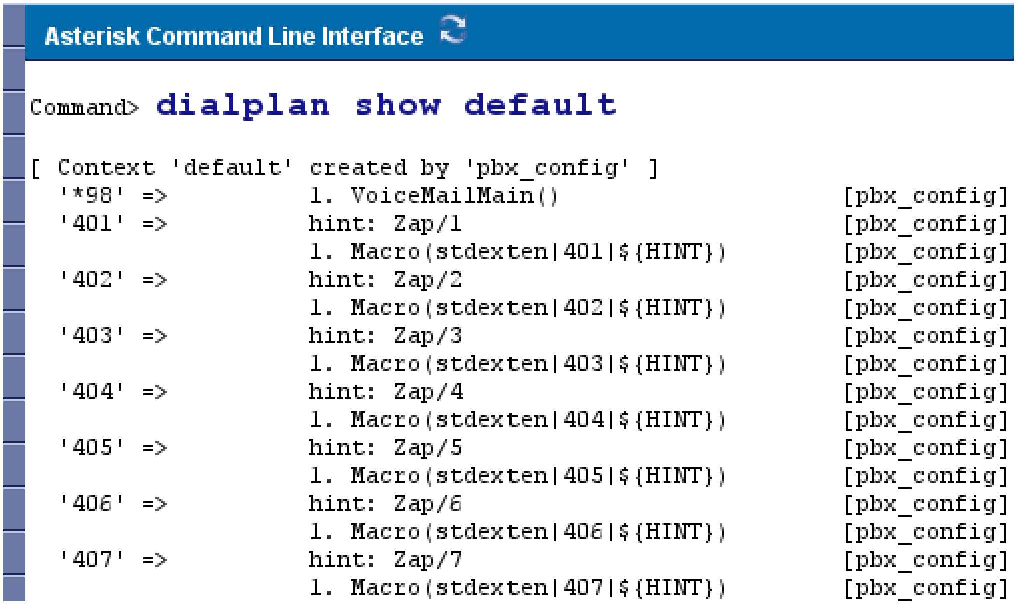
Incoming Calls

Incoming Call Rules

1	Route all unmatched incoming calls from provider 'Ports 13,14' to '402 -- FXS 2'	Edit Delete
2	Route all unmatched incoming calls from provider 'Custom - MySIPProvider' to '401 -- FXS 1'	Edit Delete

```
[DID_trunk_1]
include = default
exten = _X.,1,Goto(default|401|1)
exten = s,1,Goto(default|401|1)

[DID_trunk_2]
include = default
exten = _X.,1,Goto(default|402|1)
exten = s,1,Goto(default|402|1)
```

Asterisk automatically creates entries for all of the PBX extensions in context [default] during run time. This is illustrated below:



```
Command> dialplan show default

[ Context 'default' created by 'pbx_config' ]
  '*98' =>          1. VoiceMailMain()                      [pbx_config]
  '401' =>          hint: Zap/1                             [pbx_config]
                    1. Macro(stdexten|401|${HINT})          [pbx_config]
  '402' =>          hint: Zap/2                             [pbx_config]
                    1. Macro(stdexten|402|${HINT})          [pbx_config]
  '403' =>          hint: Zap/3                             [pbx_config]
                    1. Macro(stdexten|403|${HINT})          [pbx_config]
  '404' =>          hint: Zap/4                             [pbx_config]
                    1. Macro(stdexten|404|${HINT})          [pbx_config]
  '405' =>          hint: Zap/5                             [pbx_config]
                    1. Macro(stdexten|405|${HINT})          [pbx_config]
  '406' =>          hint: Zap/6                             [pbx_config]
                    1. Macro(stdexten|406|${HINT})          [pbx_config]
  '407' =>          hint: Zap/7                             [pbx_config]
                    1. Macro(stdexten|407|${HINT})          [pbx_config]
```

XR1000 Configuration Tips

XR1000 runs Asterisk Open Source PBX software which is extremely flexible. However, not all the customization possibilities are available via the browser-based configurator. This section contains information that will help you implement or customize functions by tweaking the Asterisk configuration files.

Recommended Process: Tweaking the Dialing Plan

When tweaking the dialing plan, we recommend following these steps:

1. Backup the current configuration (see “Backup and Restore” on page 27).
2. Make changes in the Asterisk configuration files. To access the Asterisk configuration files either:

   Use the File Editor
   See “File Editor” on page 69

   Use an SSH Terminal
   Connect to the XR1000 using an SSH terminal client. Then, use Linux vi editor for changing configuration files that are located at folder /etc/asterisk.
3. Click **Activate Changes** to apply the changes (see the explanation on p. 35).
4. Test the system.
5. Click **Save Configuration** to save the changes on the flash disk (see the warning on p. 8).
6. Backup the new configuration.

Deleting all Content of a Specific Voicemail Box

Before a user can delete the content of a specific voicemail box, the appropriate files are tweaked. After they are tweaked, deleting is a two step process.

After Tweaking

1. Dial *97. Asterisk authenticates the user by password. Following password authentication, Asterisk asks for the voicemail box number to be cleaned up.
2. Dial the number of the voicemail box. Asterisk plays back the voicemail box number and ask for confirmation. Following confirmation, Asterisk deletes all the voice messages.

Note: *You can define a phone number other than *97 and change the authentication password 1234 (see command Authenticate below).*

Tweaking that Enables Deleting Voicemail Box Content

1. Create context [vmbox-cleanup] in the file `extensions.conf`:

```
[vmbox-cleanup]
exten=*97,1,Answer
exten=*97,n,Authenticate(1234|j) ; <Change the password here>
exten=*97,n,Playback(vm-login)
exten=*97,n,Read(mbox||3|||10)
exten=*97,n,Playback(record/mailbox)
exten=*97,n,SayDigits(${mbox})
exten=*97,n,Playback(record/willBeCleanedUp)
exten=*97,n,Read(answer||1|||10)
exten=*97,n,Gotoif($["${answer}" = "1"]?delete:cancel)
exten=*97,n(delete),System(rm -rf /mnt/vm/default/"${mbox}")
exten=*97,n,Playback(record/mailboxCleanedUp)
exten=*97,n,Hangup
exten=*97,n(cancel),Playback(record/canceled)
exten=*97,n,Hangup
exten=*97,n+101,Hangup
```

2. Add the context `vmbox-cleanup` to the corresponding dial plan context. For example, by default AsteriskGUI creates DialPlan1 and the corresponding context is `numberplan-custom-1`. Adding the new context looks like this:

```
[numberplan-custom-1]
include = vmbox-cleanup
.....
```

3. On the Record Menu screen (see “Record a Menu” on page 66), record the following files:

File name	Suggested Text
mailbox	“Mailbox?”
willBeCleanedUp	"will be cleaned up. Press 1 for confirmation or 2 for cancel."
mailboxCleanedUp	"the mailbox has be cleaned up"
canceled	“canceled.”

Defining a Voicemail Box Number Different than the Extension Number

This is necessary if, for example, voice mail messages for extensions 401, 402 and 403 must be forwarded to voice mail box 401. The other extensions must not be affected.

1. Create context [define-vmbox] in file `extensions.conf`:

```
[define-vmbox]
exten => s,1,Set(VMbox=${ARG1})
exten => s,n,Set(VMbox=${IF($["${ARG1}"="402"]?"401":${VMbox})})
exten => s,n,Set(VMbox=${IF($["${ARG1}"="403"]?"401":${VMbox})})
exten => s,n,Return
```

2. In the existing context `[macro-stdexten]` in file `extensions.conf`, replace argument `${ARG1}` with `${VMbox}` in all calls to the Voicemail() function and add call to define-vmbox subroutine at the beginning. Finally you should get the following (the changes are presented as bold text):

```
[macro-stdexten];
exten => s,1,Gosub(define-vmbox,s,1)
exten => s,2,Dial(${ARG2},20)
exten => s,3,Goto(s-${DIALSTATUS},1)
exten => s-NOANSWER,1,Voicemail(${VMbox},u)
exten => s-NOANSWER,2,Hangup
exten => s-BUSY,1,Voicemail(${VMbox},b)
exten => s-BUSY,2,Hangup
exten => _s-.,1,Goto(s-NOANSWER,1)
exten => a,1,VoicemailMain(${VMbox})
```

Defining Shortcut Numbers

Many users want to use three or four digits numbers for dialing frequently used phone numbers. These numbers are called “shortcut numbers” or simply “shortcuts”.

In this example, there are two defined Service Providers:

- Ports 13,14
  This includes the two FXO lines used for domestic calls.
- MySIPProvider
  This provider is used for VoIP calls to the United Kingdom.

We need to define the following shortcuts:

Shortcut number	Full number	Service Provider
501 (to call Alice)	07712345678	“Ports 13,14” (the FXO ports)
502 (to call Bob)	07787654321	“Ports 13,14” (the FXO ports)
503 (to call my business partner in London)	442081234567	“MySIPProvider”

1. Open file `users.conf`.
2. Find the names of the sections where provider “Ports 13,14” and provider “MySIPProvider”are defined. Usually, the sections are named as trunk_n where n is a digit. For example:

```
[trunk_1]
callerid=asreceived
context=DID_trunk_1
.....
trunkname=Ports 13,14
trunkstyle=analog
zapchan=13,14

[trunk_2]
allow=all
context=DID_trunk_2
...........
trunkname=Custom - MySIPProvider
trunkstyle=customvoip
```

3. Define the context `[custom-shortcuts]` in the file `extensions.conf` and define the dialing rules. An example follows:

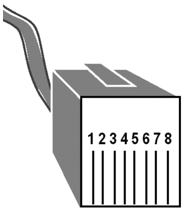
```
[custom-shortcuts]
exten = 501,1,Macro(trunkdial,${trunk_1}/07712345678)
exten = 502,1,Macro(trunkdial,${trunk_1}/07787654321)
exten = 503,1,Macro(trunkdial,${trunk_2}/442081234567)
```

4. Add context `custom-shortcuts` to the corresponding dial plan context(s). For example, by default the Asterisk GUI creates DialPlan1 and the corresponding context is `numberplan-custom-1`:

```
[numberplan-custom-1]
include = custom-shortcuts
.....
```

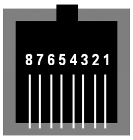
If the shortcuts are to be available for external callers, then add context `custom-shortcuts` to context `default` instead of `numberplan-custom-1`.

Cables Pin Out



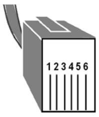
RJ-45 8 Position
Modular Plug

RJ-45 8 Position
Modular Port



RJ-45 Pin	Auxiliary Output Port	Auxiliary Input Port
1	Relay 1 Common	Input 1
2	Relay 1 Normally Open	Ground
3	Relay 1 Normally Closed	Input 3
4	Not Connected	Ground
5	Not Connected	Input 2
6	Relay 2 Common	Ground
7	Relay 2 Normally Open	Input 4
8	Relay 2 Normally Closed	Ground

RJ-11 Pin	Function	RJ-11 Pin	Function
1	Not Connected	4	Tip
2	Not Connected	5	Not Connected
3	Ring	6	Not Connected



RJ-11 6 Position Plug

RJ-11 6 Position Port