

Hybrid PBX and VoIP Gateway with (4) or (12) FXOs

Administrator's Guide

Enables your office to communicate through both telephone and VoIP lines.



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1. Todas las instrucciones de seguridad y operación deberán ser leídas antes de que el aparato eléctrico sea operado.
2. Las instrucciones de seguridad y operación deberán ser guardadas para referencia futura.
3. Todas las advertencias en el aparato eléctrico y en sus instrucciones de operación deben ser respetadas.
4. Todas las instrucciones de operación y uso deben ser seguidas.
5. El aparato eléctrico no deberá ser usado cerca del agua—por ejemplo, cerca de la tina de baño, lavabo, sótano mojado o cerca de una alberca, etc.
6. El aparato eléctrico debe ser usado únicamente con carritos o pedestales que sean recomendados por el fabricante.
7. El aparato eléctrico debe ser montado a la pared o al techo sólo como sea recomendado por el fabricante.
8. Servicio—El usuario no debe intentar dar servicio al equipo eléctrico más allá de lo descrito en las instrucciones de operación. Todo otro servicio deberá ser referido a personal de servicio calificado.
9. El aparato eléctrico debe ser situado de tal manera que su posición no interfiera su uso. La colocación del aparato eléctrico sobre una cama, sofá, alfombra o superficie similar puede bloquear la ventilación, no se debe colocar en libreros o gabinetes que impidan el flujo de aire por los orificios de ventilación.
10. El equipo eléctrico debe ser situado fuera del alcance de fuentes de calor como radiadores, registros de calor, estufas u otros aparatos (incluyendo amplificadores) que producen calor.
11. El aparato eléctrico deberá ser conectado a una fuente de poder sólo del tipo descrito en el instructivo de operación, o como se indique en el aparato.
12. Precaución debe ser tomada de tal manera que la tierra física y la polarización del equipo no sea eliminada.
13. Los cables de la fuente de poder deben ser guiados de tal manera que no sean pisados ni pellizcados por objetos colocados sobre o contra ellos, poniendo particular atención a los contactos y receptáculos donde salen del aparato.
14. El equipo eléctrico debe ser limpiado únicamente de acuerdo a las recomendaciones del fabricante.
15. En caso de existir, una antena externa deberá ser localizada lejos de las líneas de energía.
16. El cable de corriente deberá ser desconectado del cuando el equipo no sea usado por un largo periodo de tiempo.
17. Cuidado debe ser tomado de tal manera que objetos líquidos no sean derramados sobre la cubierta u orificios de ventilación.
18. Servicio por personal calificado deberá ser provisto cuando:
 - A: El cable de poder o el contacto ha sido dañado; u
 - B: Objetos han caído o líquido ha sido derramado dentro del aparato; o
 - C: El aparato ha sido expuesto a la lluvia; o
 - D: El aparato parece no operar normalmente o muestra un cambio en su desempeño; o
 - E: El aparato ha sido tirado o su cubierta ha sido dañada.

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1. Introduction

1.1 About this Guide

This guide explains how to configure and operate a Hybrid PBX and VoIP Gateway. For installation instructions, refer to the Hybrid PBX and VoIP Gateway with (4) FXO or (12) FXO (IPBX424 or IPBX1224) Installation Guide.

Products Covered

This guide explains how to configure and operate the following products:

- Hybrid PBX and VoIP Gateway with (4) FXO (IPBX424)
- Hybrid PBX and VoIP Gateway with (12) FXO (IPBX1224)

1.2 Important Terms and Conventions

1.2.1 Terms

The following terms are used in this guide.

Table 1-1. Terms and their descriptions.

Term	Description
Hybrid PBX and VoIP Gateway	Refers to a Hybrid PBX and VoIP Gateway with (4) FXO (IPBX424) or (12) FXO (IPBX1224)
Management computer	The computer that connects to the Hybrid PBX and VoIP Gateway Web-based command center to perform configuration and management tasks.

1.2.2 Conventions

Command center

This guide uses the following conventions when directing you to interact with the command center user interface.

Table 1-2. Command center conventions.

Example directions in this guide	What to do in the user interface
Select "PBX > Phone> Phone Provisioning."	On the main menu, select "PBX," then select "Phone" on the drop-down menu and then select "Phone Provisioning."
For Password, specify "secret22."	In the Password field, enter the text "secret22" exactly as shown.

1.2.3 Warnings and Cautions

It is important that you heed all warnings and cautions. They are identified as follows:

Table 1-3. Warnings and Cautions.

WARNING:	Failure to follow a warning may cause injury, death, and/or hardware damage.
CAUTION:	Failure to follow a caution may cause loss of data and/or configuration information, or cause equipment inoperability and/or security concerns.

1.3 Related Documentation

The following related documents are available at <ftp://ftp.blackbox.com/anonymous/manuals/l/>

Table 1-4. Available documentation.

Document	File Location
Hybrid PBX and VoIP Gateway Administrator's Guide (the manual you are reading now)	ftp://ftp.blackbox.com/anonymous/manuals/l/ipbx424_admin_rev1.pdf
Hybrid PBX and VoIP Gateway Installation Guide	ftp://ftp.blackbox.com/anonymous/manuals/l/ipbx424_install_rev1.pdf

2. Using the Command Center

To configure, control, and monitor the Hybrid PBX and VoIP Gateway, you must login to its Web-based command center. This requires a computer with an Ethernet port and a Web browser. Black Box recommends Mozilla Firefox® to ensure complete compatibility with the command center. Other browsers may also work.

2.1 Starting the Command Center

To reach the command center, connect your computer to the Hybrid PBX and VoIP Gateway as explained in the Hybrid PBX and VoIP Gateway Installation Guide.

1. Start your Web browser, specify `http://192.168.1.2` in the address bar, and press Enter. If you changed the IP address during installation, specify the new address instead.

(To retrieve the IP address by phone: Plug an analog phone into port FXS1 or FXS2 and dial: `##**`. Then, when prompted for a password, dial: `1234567890#`)

The Login page opens.

NOTE: If this is the first time you are starting the command center, or after a factory reset, you will see the Setup Wizard instead of the Login page. Refer to the Hybrid PBX and VoIP Gateway Installation Guide for instructions on using the wizard.



Figure 2-1. Command center.

Chapter 2: Using the Command Center

2. Specify the User Name and Password you set during installation.
3. Select a language.
4. Click on the “ENTER” button.
5. The Status page opens. See Section 2.7.

2.2 Basic Command Center Operations

The main features of the command center are: main menu, submenus, apply configuration button, and logout link.

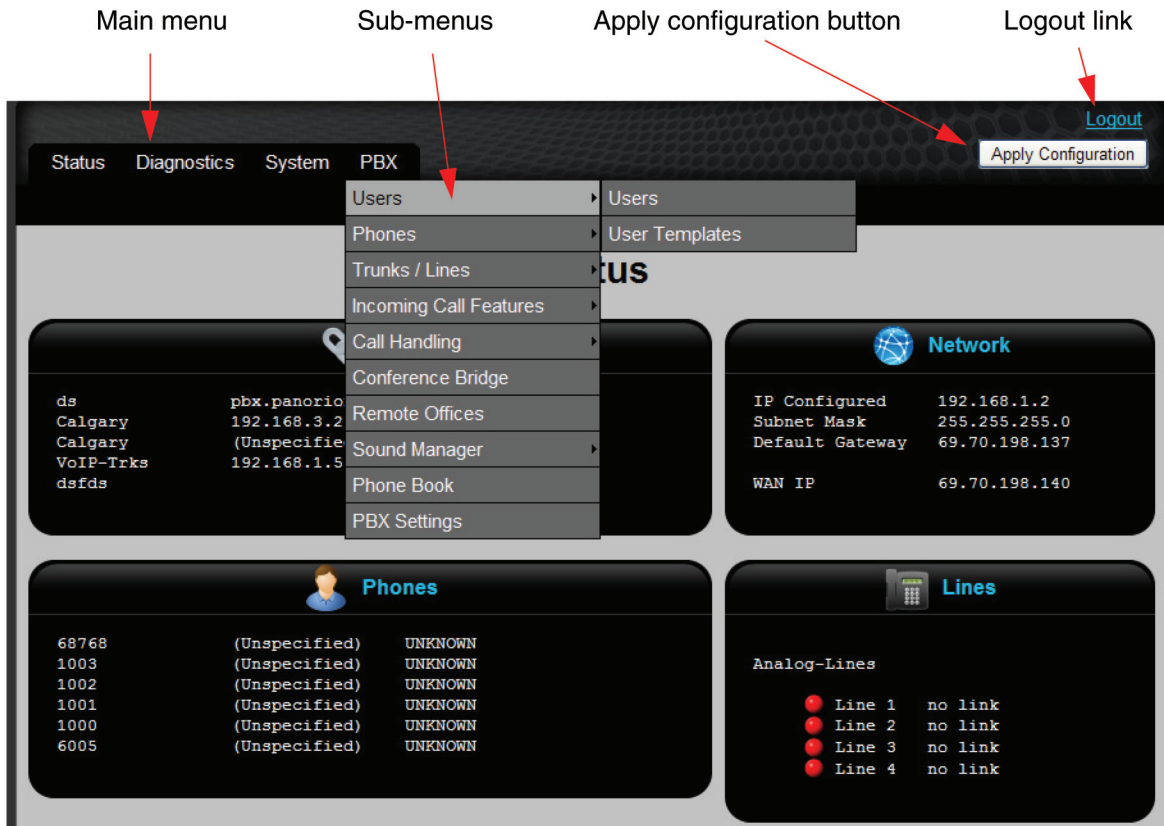


Figure 2-2. Command center features.

Applying new configuration settings

When you change a configuration setting in command center, you need to do two things to activate your change: click on “Save” and then click on “Apply.”

- When you click “Save” on a configuration page, all settings are stored in a working configuration file. These settings are not activated on the Hybrid PBX and VoIP Gateway.

If you have a number of configuration changes to make, you can keep adding them to the working configuration file by clicking “Save” on each page. When you are done, you can then apply them to the active configuration.

- When you click “Apply Configuration” (in the top right corner of the interface), the settings in the working configuration file are copied to the active configuration file on the Hybrid PBX and VoIP Gateway.

NOTE: We recommend that you only apply configuration changes during off hours when the Hybrid PBX and VoIP Gateway is not in use to avoid any service interruptions.

1. When you click “Apply Configuration,” you will be prompted to proceed.

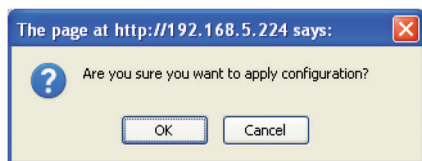


Figure 2-3. Apply configuration prompt.

2. If you click "OK," the configuration settings are applied. When done, a confirmation message is displayed indicating success or failure. If the configuration was successfully applied, click "OK" to continue.

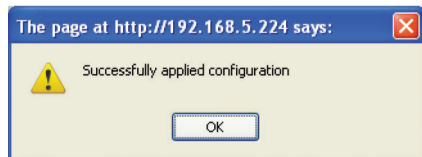


Figure 2-4. Confirmation message.

3. If there was a configuration error, you will see a list of error messages that describe what went wrong. For example, the following messages indicate that there are undefined accounting greetings, and an undefined outgoing line group called Analog.

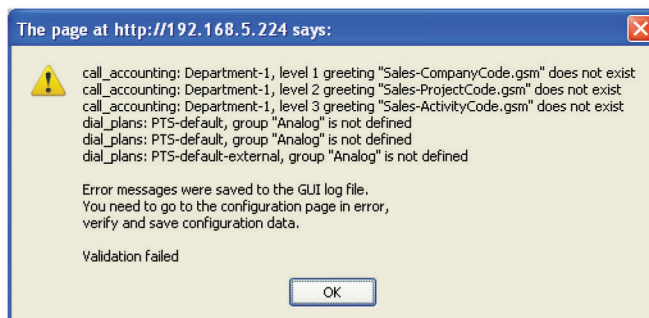


Figure 2-5. Configuration error screen.

When you click "OK," the Logs page opens showing the same messages as a reference to help you resolve the issues.

For example:

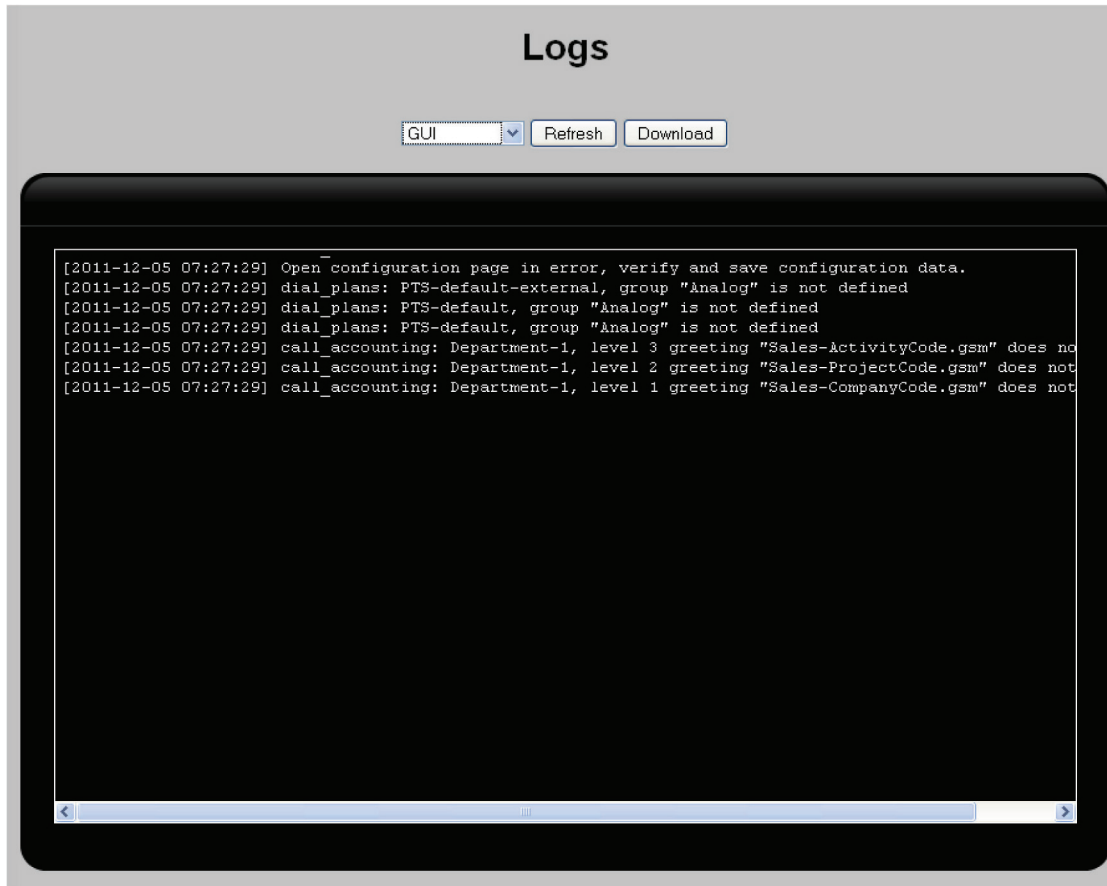


Figure 2-6. Logs page.

To return to this page, select **Diagnostics > Logs**.

2.3 Automatic Configuration Backups

Each time you click "Apply Configuration," a time-stamped backup file called lkgc (last known good configuration) is updated. To undo changes in the working configuration that have not yet been applied, you can restore the Hybrid PBX and VoIP Gateway configuration settings from this file by selecting "System > Maintenance > Backups." See Section 10.5.

2.4 Logout Link

Click the Logout link in the top right corner of the interface to exit and close command center. Any changes that were made to the working configuration that have not yet been applied will be lost.

2.5 Connecting to the Command Center via the WAN Port

In some cases, it may be useful to connect to the command center via the WAN port. To do this, you must enable the HTTP option on the Firewall page. See Section 3.4.

2.6 Configuring the Administrator Password

Select **System > Administrator Account**. The Administrator Account page opens.



Administrator Account

Please create a name and password for the administrator of this system. The administrator will be able to add and modify all parameters on the system so it is wise to choose an unknown name and password and not to share the details.

Name:

Password:

Verify Password:

Figure 2-7. Administrator account page.

Configure settings as follows:

- **Name:** Specify a name for the administrator account.
- **Password/Verify password:** To safeguard access to the administrator account we strongly recommend that you define a password that is:
 - at least twelve characters long
 - contains a combination of uppercase/lowercase letters and numbers
 - contains at least one symbol

To reset a user's voicemail password

If a user forgets their password, you can reset it as follows:

1. Select "PBX > Users > Users."
2. Click on the "Edit" button for the user's account.
3. Click "Reset Voicemail Password." The password will be reset to "1234."

2.7 Viewing PBX Status Information

The Status page shows the current status of the Hybrid PBX and VoIP Gateway. For example:



Figure 2-8. Status screen.

2.7.1 VoIP

If you have VoIP lines, check the VoIP box to verify their status. For each line you should see an IP address and a status value of OK (xx ms), where xx indicates how long it takes the Hybrid PBX and VoIP Gateway to communicate with the VoIP carrier.

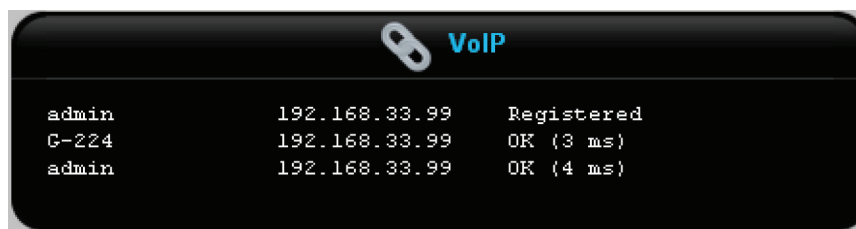


Figure 2-9. VoIP screen.

2.7.2 Status Messages

- Registered: Indicates a valid connection to a VoIP provider or remote office.
- Unregistered: No connection to a VoIP provider or remote office.
- Timeout: Registration to the VoIP provider or remote office timed out.
- Rejected: Invalid credentials (username, password) were used to connect to the VoIP provider or remote office.
- Failed: Registration failed after several retries.
- AuthSent: Indicates an attempt was made to authenticate with the VoIP provider or remote office.
- UNREACHABLE: The VoIP provider or remote office is unreachable. (This occurs if the Hybrid PBX and VoIP Gateway does not receive a reply to a SIP OPTIONS packet within two seconds.)
- OK (xxx ms): The VoIP provider or remote office was reached in xxx milliseconds.
- Lagged: The VoIP provider or remote office was reached, but not within the default timeout.
- Unmonitored: Monitoring of the VoIP provider or remote office has been disabled. The value of Qualify is set to "no." See Section 4.3.

2.7.3 Lines

Check the "Lines" box to verify the status of your analog phone lines. For each line you should see a green light and a status value of "connected."

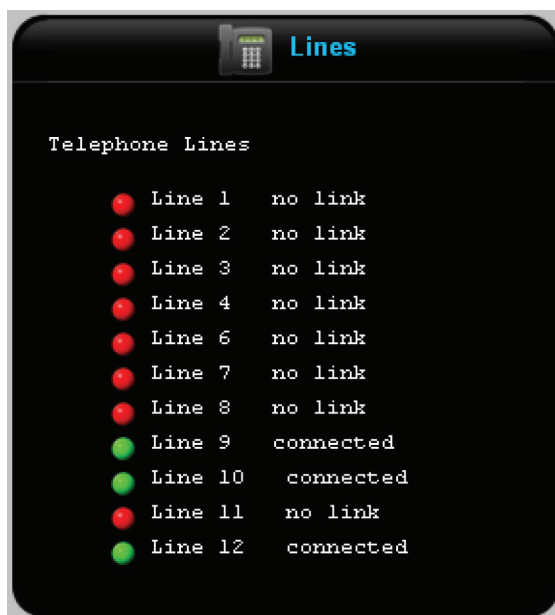


Figure 2-10. Lines screen.

Chapter 2: Using the Command Center

Possible status values are:

- connected: The line is physically connected and operational.
- no link: The line is not connected or has failed.

2.7.4 Network

Shows the addresses assigned to the LAN and WAN ports (if defined).

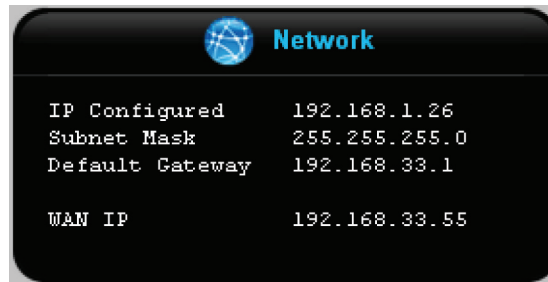


Figure 2-11. LAN and WAN network addresses.

- IP Configured: Address assigned to the LAN ports.
- Subnet Mask: Subnet mask for the LAN ports IP address.
- Default Gateway: IP address of the default gateway. If the WAN port is defined, the default gateway will be on the WAN network.
- WAN IP: Address assigned to Ethernet port 4.

To see if your connection to the Internet is working:

1. Select "Diagnostics > Diagnostics > Ping."

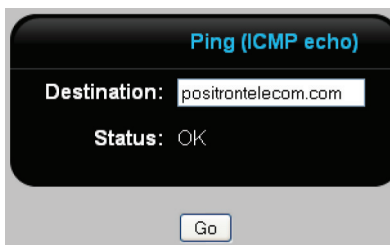


Figure 2-12. Ping screen.

2. Specify the address of a site on the Internet. For example: blackbox.com.
 3. Click "Go."
- If Status displays "OK," then the connection is working.
 - If Status displays "No response," then select System > Network > Network and make sure that the Default Gateway and DNS server are set to the correct values.

2.7.5 Storage

- For a USB flash drive you should see memory totals.

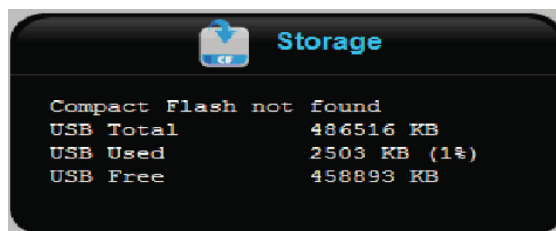


Figure 2-13. Storage screen for USB flash drive.

2.7.6 Phones

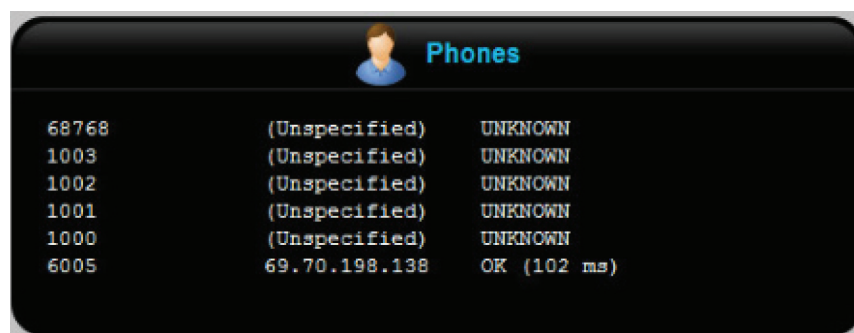


Figure 2-14. Phones screen.

Status messages include:

- OK (xxx ms): Phone can be reached in xxx milliseconds.
- UNREACHABLE: The SIP phone is unreachable. (This occurs if the Hybrid PBX and VoIP Gateway does not receive a reply to a SIP OPTIONS packet within a define interval. (See Qualify in Section 6.4.2.)
- Lagged: The VoIP provider was reached, but not within the default timeout.
- Unmonitored: Monitoring of the phone has been disabled. The value of Qualify is set to "no." (See Qualify in Section 6.4.2.)

3. Network Configuration

3.1 Deployment Strategies

The following topics describe key points to consider when deciding how to integrate the Hybrid PBX and VoIP Gateway into your network, or when building a new deployment from the ground up.

Refer to scenarios on the following pages to see how these issues are addressed using different deployment strategies.

3.1.1 Automatic Phone Provisioning and DHCP Option 66

For most installations, we recommend that the Hybrid PBX and VoIP Gateway acts as the DHCP server for your network. The main reason for this is to support the automatic provisioning of SIP phones. Provisioning works as follows: at startup, SIP phones retrieve a configuration file from a server (like the Hybrid PBX and VoIP Gateway), that defines operational settings. To do this, the phones must know the IP address of the server.

You can program this address into the phones manually, or you can automatically distribute the address using a DHCP server that supports option 66. Not all DHCP servers allow you to define this option. The DHCP server on the Hybrid PBX and VoIP Gateway does, without requiring any configuration. So, this is why the easiest installation method is to disable your existing DHCP server and use the one on the Hybrid PBX and VoIP Gateway.

(If your DHCP server supports option 66, and you want to use it instead of the Hybrid PBX and VoIP Gateway, see “Configuring a Third-Party DHCP Server” in Section 3.3.

3.1.2 Assigning the Gateway for Internet Traffic

The Hybrid PBX and VoIP Gateway is a VoIP gateway. It only forwards SIP and voice traffic between its LAN and WAN ports. Therefore, it cannot be used as a gateway to the Internet for the computers on your network or for SIP phones that have Internet capabilities. This means that when you configure the Hybrid PBX and VoIP Gateway DHCP server, it should not return its own address as the default gateway or DNS server. You need to adjust these settings to point to the router on your network that provides access to the Internet.

3.1.3 Port Forwarding, Public IP Addresses, and Security Issues

If the connection to the Internet/VoIP provider is behind a router (with NAT enabled) or firewall, direct access to the VoIP lines you need to install on the Hybrid PBX and VoIP Gateway will be blocked. To allow access, you will have to configure port forwarding on the router and/or open ports on the firewall. By default, you need to allow UDP ports 5060 and 10000 to 20000 to reach the Hybrid PBX and VoIP Gateway. See “Step 3: Configure port forwarding” in Section 4.2.

To avoid these issues, you can set up your network to allow for a public IP address to be assigned to the WAN port on the Hybrid PBX and VoIP Gateway. The WAN port is protected by a firewall, and an automatic port blocking feature, so exposing it to the Internet is safe.

- See Section 3.4.
- See Section 3.5.

3.1.4 VoIP Bandwidth Requirements

Your total bandwidth requirements are directly related to the number of VoIP lines you intend to use.

Each VoIP line requires approximately 87.2 kbps on a Hybrid PBX and VoIP Gateway.

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Scenario 2

In this scenario, the Hybrid PBX and VoIP Gateway WAN port is directly connected to a switch provided by the Internet ISP. Traffic from the SIP phones (green line) reaches the Hybrid PBX and VoIP Gateway on its LAN port and is then sent to the VoIP provider via the WAN port. One advantage of this setup is that the WAN port gets a public IP address on the Internet. This makes it easy to configure VoIP support and provide access for remote phone, as port forwarding does not have to be configured on the Gateway/Router as in scenario 1.

As in scenario 1, the Hybrid PBX and VoIP Gateway acts as the DHCP for the network, allowing it to automatically provision the SIP phones. However, to support Internet access (yellow line), it assigns the Gateway/Router as the DNS server and default gateway for all network devices.

(If your existing DHCP server supports option 66, you can continue to use it if you make configuration adjustments. For details, see “Configuring a third-party DHCP server” in Section 3.3.)

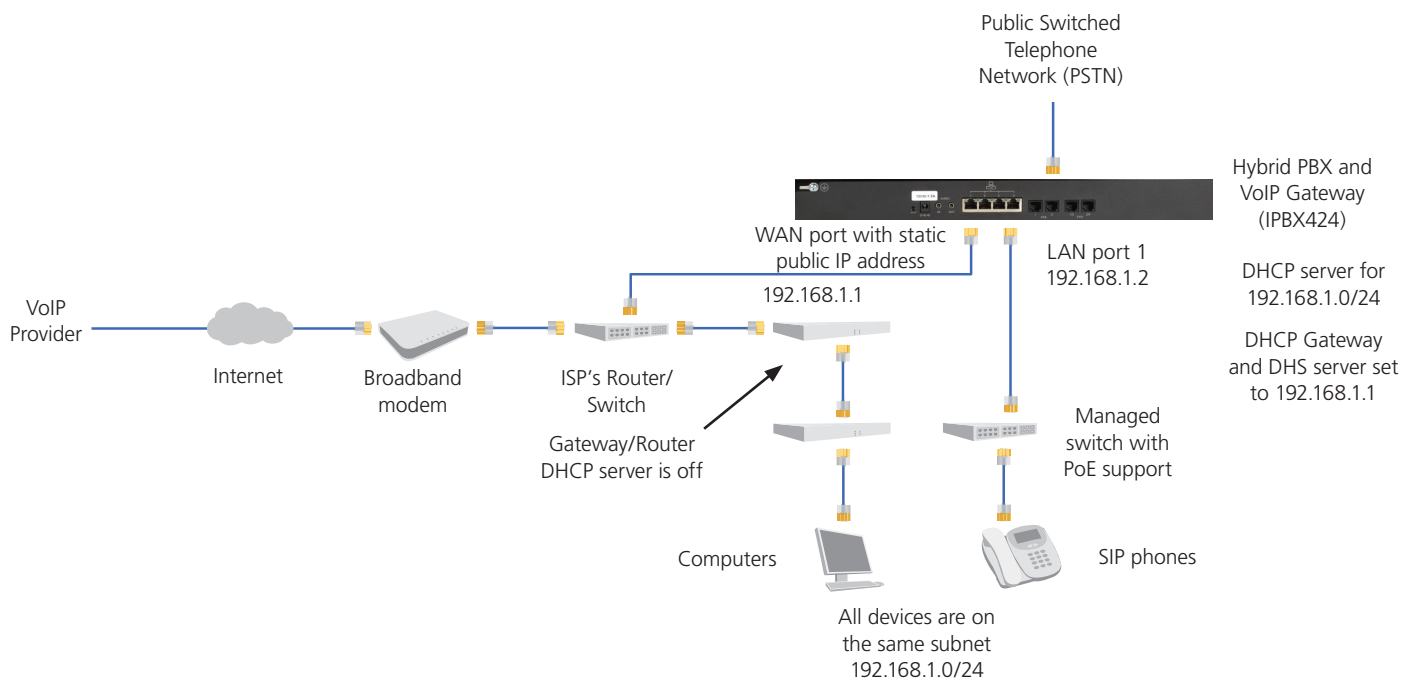


Figure 3-2. SIP Gateway WAN port directly connected to a switch provided by the ISP.

Scenario 3

In this scenario, voice (green line) and data traffic (yellow line) are each carried on its own network. This requires that both the Gateway/Router and the Hybrid PBX and VoIP Gateway are configured as DHCP servers, each one handling a different subnet. In this setup, the WAN port on the Hybrid PBX and VoIP Gateway gets a public IP address on the Internet. This makes it easy to configure VoIP lines and provide support for remote phones, as port forwarding does not have to be configured.

If the SIP phones you are installing do not require Internet access, then you can simplify this installation by removing the Router and the connection between the two networks.

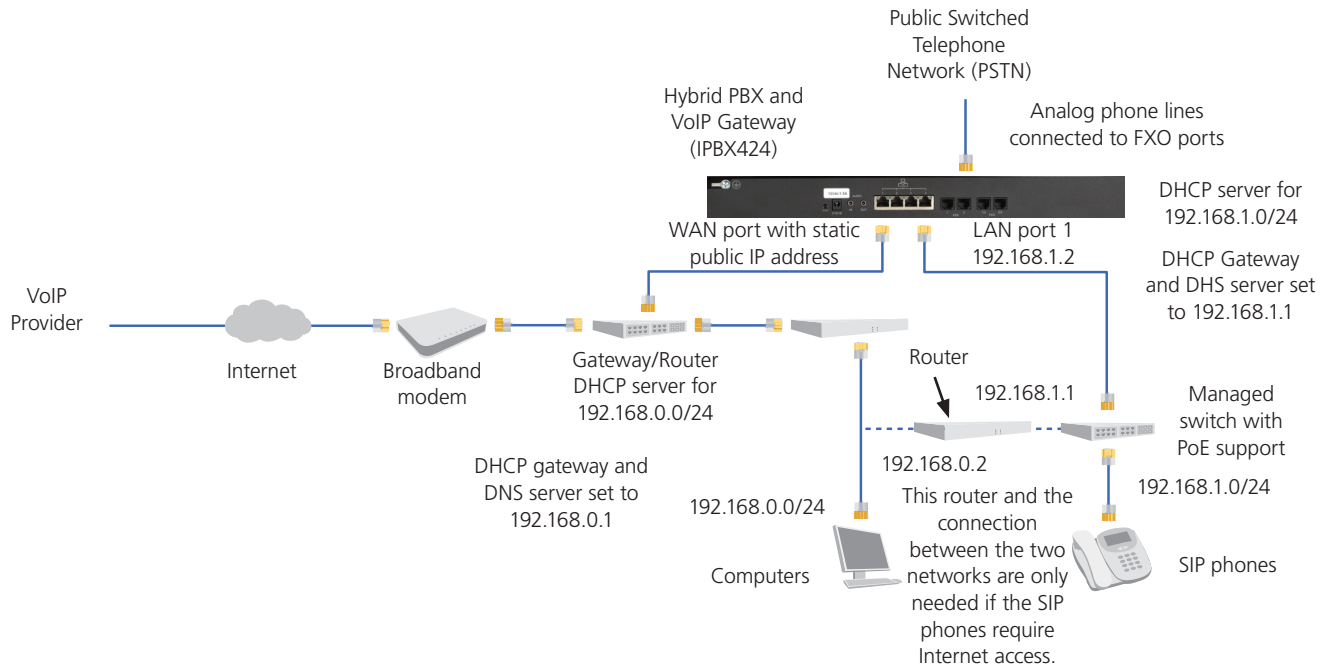


Figure 3-3. Voice and data traffic each carried on their own network.

Chapter 3: Network Configuration

Scenario 4

This is a variation of scenario 3, in which the Hybrid PBX and VoIP Gateway is connected to the Gateway/Router instead of a switch. This means the IP address assigned to the WAN port is not public on the Internet, requiring port forwarding to be configured on the Gateway/Router.

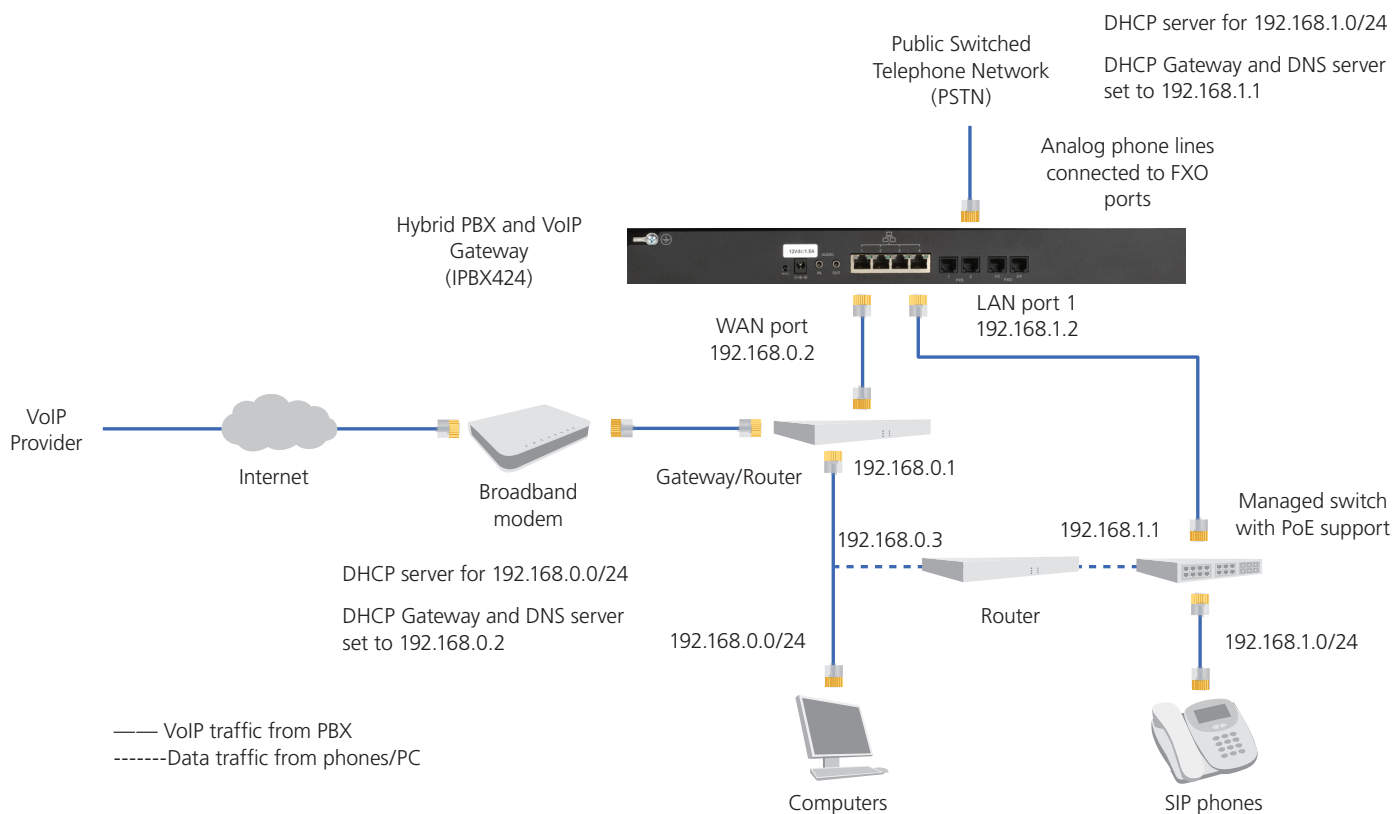


Figure 3-4. Hybrid PBX and VoIP Gateway connected to the Gateway/Router.

Scenario 5

In this standalone installation, the Hybrid PBX and VoIP Gateway is directly connected to the Internet through its own broadband connection. Since the WAN port has a public IP address on the Internet, connection to the VoIP provider is simplified, as is allowing remote users to connect. (In this scenario, SIP phones are not able to access the Internet for web browsing, etc.)

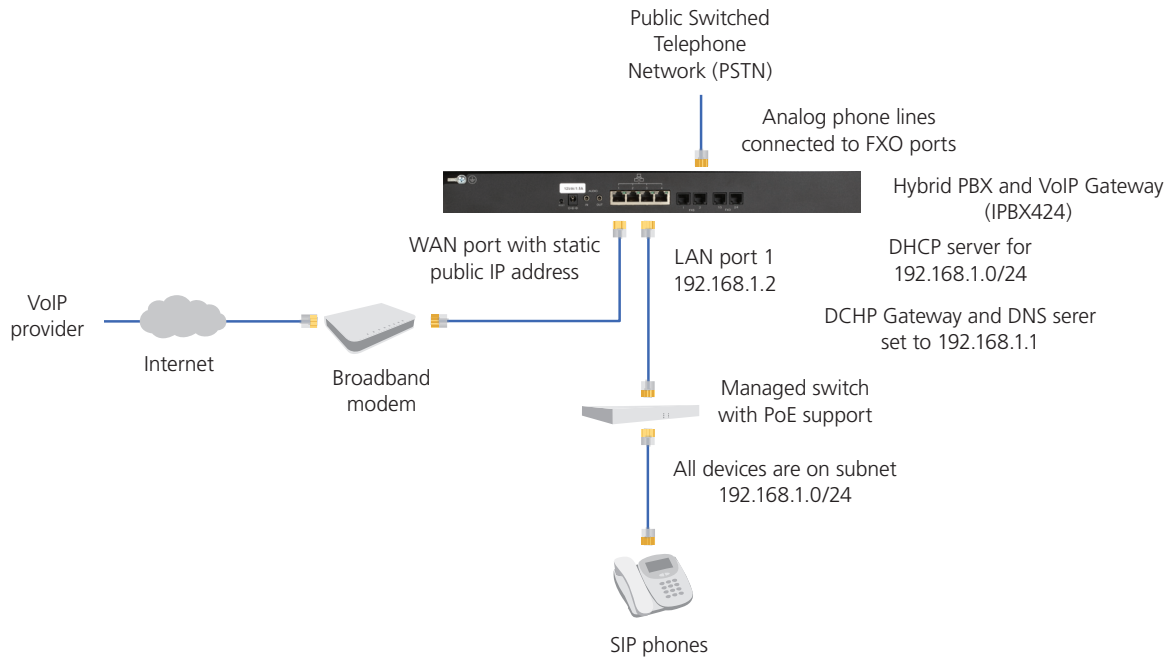


Figure 3-5. Hybrid PBX and VoIP Gateway directly connected to the Internet through its own broadband connection.

3.2 Configuring the LAN and WAN Ports

To configure networking settings for the LAN and WAN interfaces, select System > Network > Network. The Network page opens.

The screenshot shows a web-based configuration interface titled "Network". At the top, there is a horizontal menu with tabs: "Network", "DHCP/NTP", "SMTP", "Firewall", "Block IP", "SIP/RTP", and "Services". The "Network" tab is selected. Below the tabs is a large black rounded rectangle containing the configuration fields. The fields are organized as follows: "Host Name:" followed by a text input field; a horizontal separator line; two columns of fields labeled "LAN" and "WAN"; "LAN" column: "IP Address:" with a text input field containing "192.168.1.2" and "Subnet Mask:" with a text input field containing "255.255.255.0"; "WAN" column: "WAN IP Address:" with an empty text input field and "WAN Subnet Mask:" with an empty text input field; another horizontal separator line; "Default Gateway:" with an empty text input field; "DNS Server:" with an empty text input field; a third horizontal separator line; "Port 3:" followed by a dropdown menu currently showing "LAN". At the bottom center of the black rectangle is an "Apply" button.

Figure 3-6. Network page.

Configure settings as follows. Click “Apply” when done.

- Host Name: Specify a name to identify the Hybrid PBX and VoIP Gateway on the network. This name can be anything meaningful to the organization and can be useful in the event that multiple Hybrid PBX and VoIP Gateways are installed.

LAN

Use these fields to assign an IP address to ports 1, 2, and 3. These ports share the same IP address.

- IP Address: Specify an IP address in dotted-decimal notation (nnn.nnn.nnn.nnn). Default: 192.168.1.2
- Subnet Mask: Specify a subnet mask that is appropriate for the address you entered. Default: 255.255.255.0

WAN

If your scenario makes use of the WAN port to connect the Hybrid PBX and VoIP Gateway to the Internet via a broadband router, configure settings as follows:

- IP address: Specify an IP address in dotted-decimal notation (nnn.nnn.nnn.nnn). By default, no address is assigned. If you are connected to the Internet via cable or DSL the router/modem, you must assign an address that is on the same subnet as the router/modem provided by your ISP or network administrator.

- Subnet Mask: Specify the subnet mask that is appropriate for the address you entered. By default, the mask is set to 255.255.255.0.
- Default Gateway: Specify the IP address of the default gateway on your network provided by your ISP or network administrator. This is the device that provides access to the Internet.
- DNS Server: Specify the IP address of the primary DNS server provided by your ISP or network administrator. This will be the address of default gateway.

Port 3: Select the functionality provided by Ethernet port 3.

- LAN: Normal operation. Bridged to ports 1 and 2.
- LAN Monitoring: Shows traffic being sent and received on Ethernet ports 1 and 2.
- WAN Monitoring: Shows traffic being sent and received on Ethernet port 4.

3.3 Configuring DHCP and NTP Settings

Select "System > Network > DHCP/NTP." The DHCP/NTP page opens.

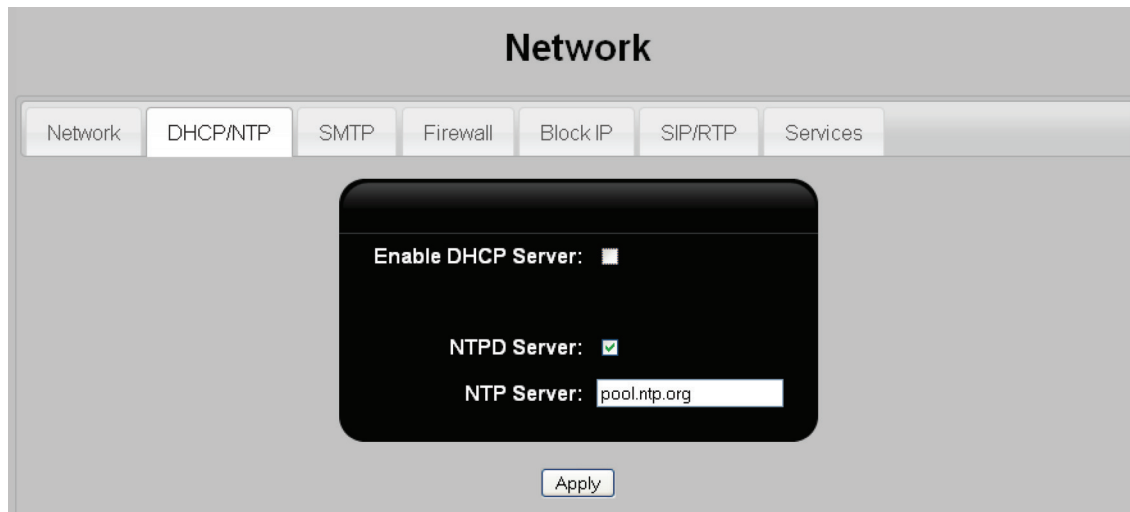


Figure 3-7. DHCP/NTP page.

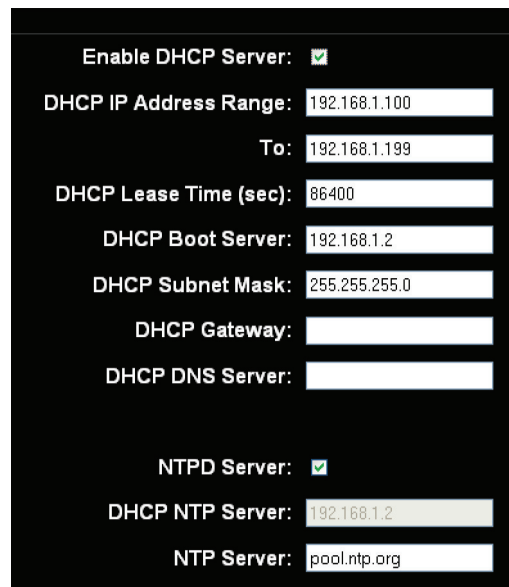
Configure settings as follows. Click "Apply" when done.

Chapter 3: Network Configuration

- **Enable DHCP Server:** For most installations, we recommend that the Hybrid PBX and VoIP Gateway acts as the DHCP server for your network. This will make it easier to manage the IP address assigned to your VoIP phones, allow for automatic phone provisioning, and ensure that the proper gateway and DNS servers are assigned. If you are connected to the Internet via cable or DSL, the broadband router/modem is generally set up as the DHCP server. Therefore, before you connect the Hybrid PBX and VoIP Gateway to the broadband router/modem, you should disable DHCP server support on this device.

Default: Disabled

- When this option is enabled, additional configuration settings are displayed.



The screenshot shows a configuration window with a dark background. At the top, 'Enable DHCP Server:' is checked with a green checkbox. Below it, 'DHCP IP Address Range:' has a text box with '192.168.1.100' and a 'To:' label followed by a text box with '192.168.1.199'. 'DHCP Lease Time (sec):' has a text box with '86400'. 'DHCP Boot Server:' has a text box with '192.168.1.2'. 'DHCP Subnet Mask:' has a text box with '255.255.255.0'. 'DHCP Gateway:' and 'DHCP DNS Server:' have empty text boxes. Further down, 'NTPD Server:' is checked with a green checkbox. 'DHCP NTP Server:' has a text box with '192.168.1.2'. 'NTP Server:' has a text box with 'pool.ntp.org'.

Figure 3-8. Additional configuration settings.

- **DHCP IP Address Range:** Specify the starting IP address for the DHCP address pool. Default: 192.168.1.100
- **To:** Specify the ending IP address for the DHCP address pool. Default: 192.168.1.199
- **DHCP Lease Time (sec):** Specify the number of seconds for which an assigned DHCP address is valid. When this time expires, DHCP clients (such as a phone) must request a new address from the DHCP server (Hybrid PBX and VoIP Gateway). Additionally, an address cannot be reused until the lease expires (unless it is explicitly released by the DHCP client). Consequently, setting a long lease time can cause the address pool to become depleted if a large number of devices connect and disconnect from the network within a time period that is shorter than the lease time. This results in leases being reserved for devices that are no longer connected to the network, potentially blocking new devices from accessing the network. Recommended settings are 120 seconds if 3G is used, and 86400 seconds if it is not used. Default: 120 seconds
- **DHCP Boot Server:** Specify the boot server IP address to be assigned to DHCP clients (SIP phones). SIP phones contact the boot server to retrieve their configuration information when starting up. Set this to the Hybrid PBX and VoIP Gateway LAN IP address.
- **DHCP Subnet Mask:** Specify the subnet mask for all DHCP addresses. Default: 255.255.255.0
- **DHCP Gateway:** Specify the IP address of the gateway that DHCP clients (SIP phones) will use to access the Internet.
- **DHCP DNS Server:** Specify the IP address of the primary DNS server that DHCP clients (SIP phones) will use.
- **NTPD Server:** When enabled, the Hybrid PBX and VoIP Gateway will set the system date and time based on values retrieved from the specified NTP Server.

- **DHCP NTP Server:** Specify the address of the NTP server that will be returned to DHCP clients (SIP phones). SIP phones can use this address to contact an NTP server to automatically set the date and time. For most installations, set this to the Hybrid PBX and VoIP Gateway LAN IP address.
- **NTP Server:** Set NTP Server to the host name of a time server. The default setting is pool.ntp.org, which is a public time server on the Internet.

Configuring a third-party DHCP server

If the Hybrid PBX and VoIP Gateway is not the DHCP server on your network, you can still use automatic phone provisioning. To do this, you need to configure the third-party DHCP server to return the IP address of the Hybrid PBX and VoIP Gateway as the value for option 66 (boot server).

The following steps show how to configure option 66 on the Microsoft 2003 DHCP Server.

(Configuration steps on other DHCP servers will be different.)

1. Open the Microsoft DHCP Server management console and select your DHCP server.
2. In the Toolbar, click "Action."
3. Select "Set Predefined Options" from the menu.
4. Click "Add" to add the DHCP options. The Option Type window opens.
5. Enter the Name. For option 66, enter "Boot Server Host Name."
6. For the Data Type, select "String."
7. For the Code, specify "66."
8. Enter a Description.
9. For option 66, specify "TFTP Boot Server Host Name."
10. Click "OK," then enter the IP address of the Hybrid PBX and VoIP Gateway for option 66.

3.4 Setting up the Firewall

The Hybrid PBX and VoIP Gateway contains an integrated firewall to protect the WAN port when it is connected to a public network like the Internet, or when a 3G device is being used.

NOTE: The firewall does not affect traffic on the LAN ports.

To configure the firewall, select "System > Network > Firewall." The Firewall page opens.

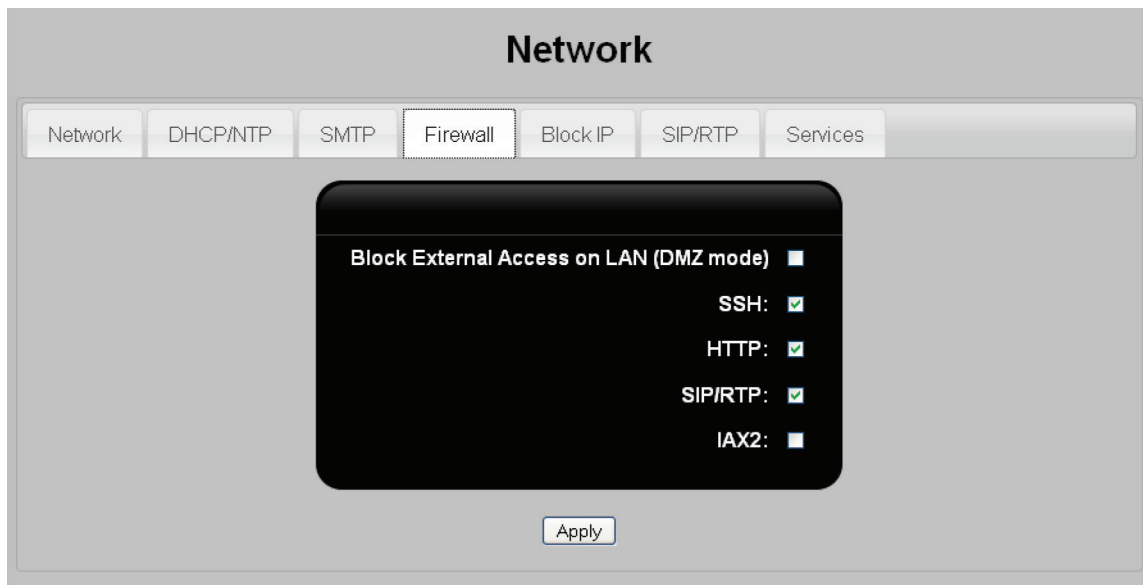


Figure 3-9.

Configure settings as follows. Click “Apply” when done.

- **Block External Access on LAN (DMZ mode):** Enable this option when the WAN IP address resides on a DMZ network (which generally exposes it to the public Internet). When enabled, the WAN port will only accept traffic from the router’s LAN port and not external devices.
- **SSH:** For troubleshooting only. Contact Black Box Technical Support at 724-746-5500 or info@blackbox.com for more information.
- **HTTP:** When enabled, access to the command center is supported via the WAN port.
- **SIP/RTP:** When enabled, allows SIP/RTP traffic to enter the WAN port. Enable this option to support remote phone connections via the WAN port, or when registering to a VoIP carrier when Internet access goes through a device that does NAT.
- **IAX2:** When enabled, allows external IAX2 traffic into the WAN port. Enable this option when your VoIP provider supports IAX2 and Internet access goes through a device that does not.

3.5 Controlling PBX Access Based on IP Address

This feature enables you to control access to the Hybrid PBX and VoIP Gateway based on a device’s IP address or network of origin. To setup blocking, select “System > Network > Block IP.” The Block IP page opens.

Network

Network DHCP/NTP SMTP Firewall **Block IP** SIP/RTP Services

Expiry: 172800

Failure attempts: 3

Permitted Networks: / lan

192.168.5.114/255.255.255.255,wan

Add Delete

Blocked Networks: /

Add Delete

PBX Blocked IPs:

Delete

Apply

Figure 3-10. Block IP page.

The Block IP feature has been designed to automatically identify malicious attempts to gain access to the Hybrid PBX and VoIP Gateway via the WAN port. If a remote phone fails to log in more than the specified number of times (as defined by Failure attempts), then the IP address of the phone is added to the PBX Blocked IPs list. The phone will remain in this list for the number of seconds defined by Expiry. To permanently block the phone, you must add its address to the Blocked Network list.

We recommend that you check the PBX Blocked IPs list on a weekly basis to ensure that you catch malicious IPs and permanently block them by adding them to Blocked networks list.

Configure settings as follows. Click “Apply” when done.

- Expiry: Blocked IPs are released after the time (in seconds) specified in this field has elapsed.
- Failure attempts: Select the number of failed login attempts before traffic is blocked from an IP address.
- Permitted Networks: Traffic will NOT be blocked from these IP addresses even if failures occur. A common use for this feature is to support remote phones dialing in via a VPN. By adding the remote VPN network to this list, users will not automatically be blocked if they fail to enter the correct username or password multiple times.

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To add an address, specify a single IP address or the address for an entire subnet in the first box. In the second box, specify the appropriate network mask, then click "Add."

- **Blocked Networks:** All traffic is blocked from IP addresses in this list. To add an address, specify a single IP address, or the address for an entire subnet in the first box. In the second box, specify the appropriate network mask, then click "Add."
- **PBX Blocked IPs:** IP addresses listed in this list are blocked due to exceeding the limit on unsuccessful attempts (as defined by Failure attempts) to connect to the Hybrid PBX and VoIP Gateway. The address will remain in this list for the number of seconds defined by Expiry.

3.6 Configuring Network Services

Select "System > Network > Services." The Services page opens.

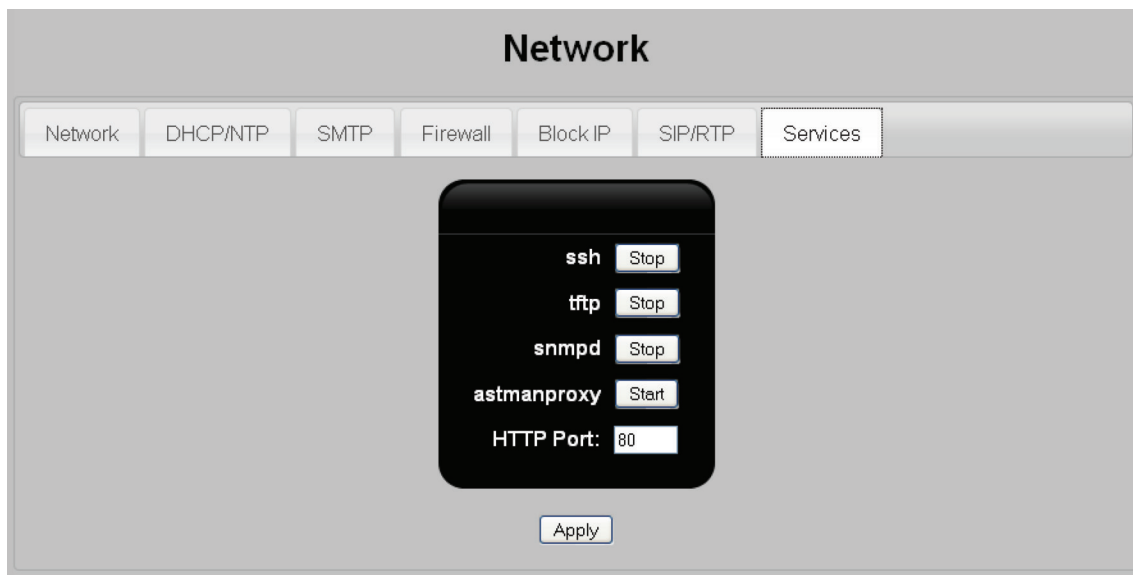


Figure 3-11. Services page.

Configure settings as follows. Click "Apply" when done.

- **ssh:** For use by Black Box Technical Support only.
- **tftp:** For future use.
- **snmpd:** For future use.
- **astmanproxy:** When enabled, allows users to make calls from Microsoft Outlook. Requires that the Outlook option is enabled in each user account (see "Outlook" in Section 6.2.1 and "Outlook" in Section 6.2.2). For information on how to configure Outlook, see the Hybrid PBX and VoIP Gateway User's Guide. Default: Stopped
- **HTTP Port:** This is the port on which you gain access to the command center using your Web browser. Default: 80

4. Setting up Trunks and Lines

4.1 Managing Trunks and Lines

The Hybrid PBX and VoIP Gateway can connect to the traditional PSTN (Public Switched Telephone Network) and to VoIP (Voice over Internet Protocol) services that are delivered over the Internet (or a private backbone network).

The connection to these services is typically called a trunk. A trunk can be composed of analog lines or a set of virtual VoIP lines (also called SIP trunks) that are delivered over a broadband data connection (DSL or cable).

- If you are going to provide analog calling to your users, you need to connect one or more analog lines to the FXO ports on the Hybrid PBX and VoIP Gateway as described in the Hybrid PBX and VoIP Gateway Installation Guide. Different models of the Hybrid PBX and VoIP Gateway support a different number of analog lines.
- To support VoIP, you need to install one or more VoIP trunks as described in the Hybrid PBX and VoIP Gateway Installation Guide.

Once your lines are installed, you are ready to configure them. If you ran the setup wizard, you may have already defined settings for your lines. If not, refer to the following sections to configure the trunks/lines.

Once your trunks/lines are configured, create outgoing line groups to help simplify the implementation of outgoing calling rules. See Section 4.7.

NOTE: Whenever you change the settings for trunks/lines you need to restart the Hybrid PBX and VoIP Gateway.

To view all trunks/lines

Select "PBX > Trunks/Lines > Trunks/Lines." The Trunks/Lines list page opens.



Figure 4-1. Trunks/Lines page.

This page lists all defined trunks/lines. The following information is presented for each trunk/line.

- Name: Name that identifies the trunk/line.
- Type: Trunk/line type: analog or voip.
- Action: Click the appropriate button to Edit or Delete a trunk/line. The Edit button displays the Trunks/Lines edit page. See Section 4.3 and Section 4.4.
- Add button: To add a new trunk/line, click Add, then see Section 4.2 or Section 4.4.

Default trunks/lines

By default, the Telephone Lines group is created and has all installed analog lines assigned to it. If you want to create additional analog line groups, then you have to first edit the Telephone Lines group and remove one or more lines from it.

4.2 Adding an SIP Trunk/Line

Most of the settings you need to configure a SIP trunk are supplied by your VoIP service provider or are already built into the default provider templates that are installed in the Hybrid PBX and VoIP Gateway. If your VoIP provider is not in the following list, you must define a custom provider template before adding the SIP trunk/line as described in this procedure. See Section 4.6.

Chapter 4: Setting Up Trunks and Lines

- babyTEL
- BroadConnect
- Dezmocom
- UnifiedVox
- Unlimited
- SIPGate
- PanOrion

Step 1: Configure the SIP trunk/line

1. Click Add on the Trunks/Lines list page. This displays the Trunks/Lines add page.

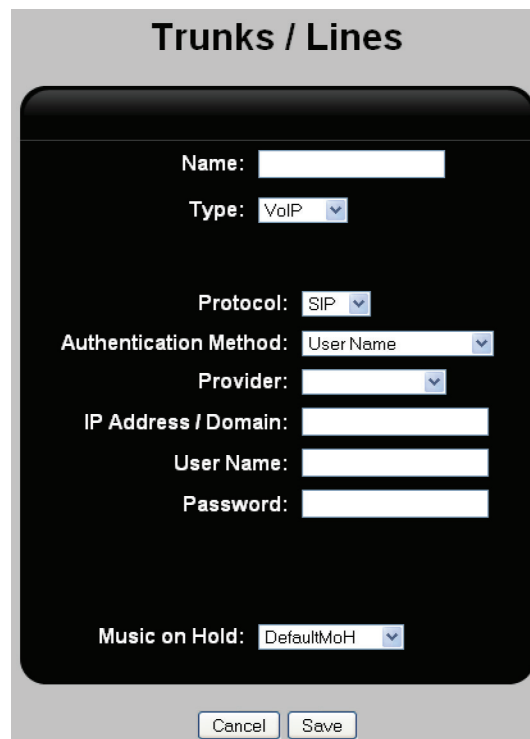


Figure 4-2. Trunks/Lines page.

2. Configure the following settings using the information from your VoIP provider.

- Name: Specify a name to identify the line on the Hybrid PBX and VoIP Gateway.
- Type: Select VoIP.
- Protocol: Select the protocol supported by the line: SIP or IAX2.
- Authentication Method: Select the method that will be used to identify the Hybrid PBX and VoIP Gateway to the VoIP provider.
- Provider: (Only available if SIP is selected for Protocol.) Select your VoIP provider. The Hybrid PBX and VoIP Gateway automatically defines configuration settings for a number of common providers. If your provider is not on this list, you will need to create a custom template for it. See Section 4.6.
- IP Address/Domain: Specify the IP address or domain name provided by your VoIP provider for your VoIP lines.
- User Name: (Only available if User Name is selected for Authentication Method.) Specify the user name assigned to you by your provider.

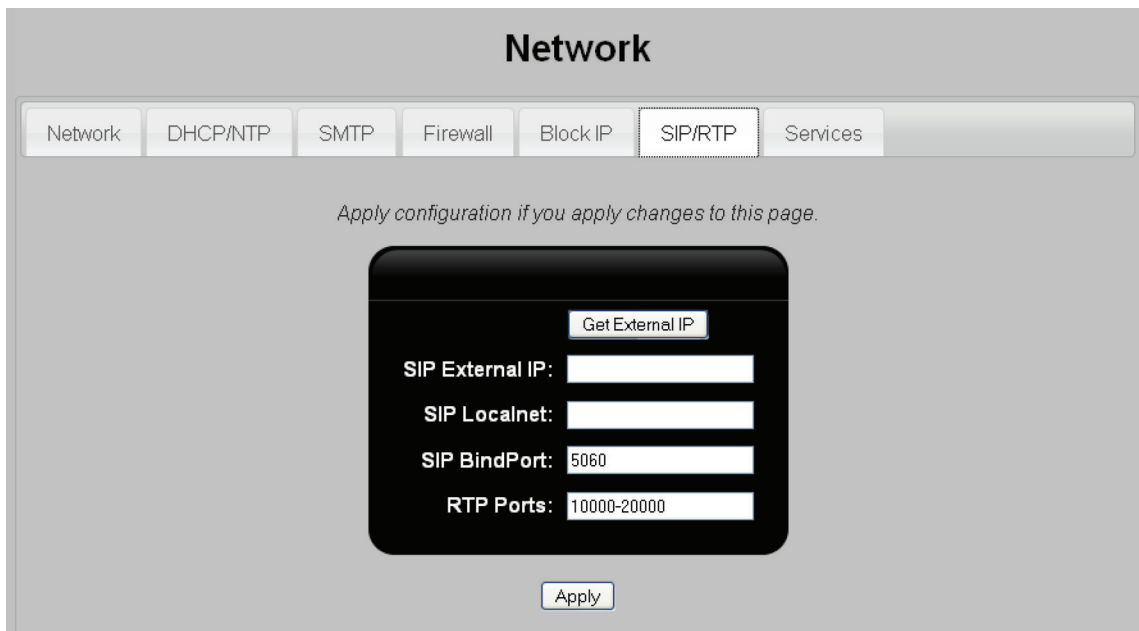
- Password: (Only available if User Name is selected for Authentication Method.) Specify the password assigned to you by your provider.
- Music on Hold: Select the method that will be used to provide music when incoming callers on the VoIP lines are put on hold. By default the following choices are available:
 - DefaultMoH: Select the default music on hold sound file.
 - DefaultAudioIn: Select the Audio In port.

For more information, see Section 6.8.

3. Click Save when you are done. The new line will be added to the Trunks/Lines list page.
4. If you want to define advanced settings for the new trunk/line, click Edit then see Section 4.3.
5. Click "Apply Configuration" and then select "System > Restart" to activate the new line/trunk.

Step 2: Configure SIP/RTP support

1. Select "System > Network > SIP/RTP." The SIP/RTP page opens.



The screenshot shows a web interface titled "Network" with a tabbed menu at the top containing "Network", "DHCP/NTP", "SMTP", "Firewall", "Block IP", "SIP/RTP" (which is selected), and "Services". Below the tabs, a message reads "Apply configuration if you apply changes to this page." In the center is a black rounded rectangle containing a "Get External IP" button, four input fields labeled "SIP External IP:", "SIP Localnet:", "SIP BindPort:" (with the value "5060"), and "RTP Ports:" (with the value "10000-20000"), and an "Apply" button at the bottom.

Figure 4-3. SIP/RTP page.

1. Click "Get External IP." This displays the public IP address that the VoIP carrier will use when communicating with the Hybrid PBX and VoIP Gateway. If the Hybrid PBX and VoIP Gateway WAN port is directly connected to a broadband modem, then this will be the address assigned to the WAN port. If the Hybrid PBX and VoIP Gateway connects to the Internet via a network connected to the LAN port, the command returns the public IP address of the router/gateway/modem making the connection.
- SIP External IP: Set this to the value returned by Get External IP. This is the public IP address that remote users will use to connect to the Hybrid PBX and VoIP Gateway.
 - SIP Localnet: If the SIP phones at your location are installed on more than one subnet, you need to identify these networks here. For example, if your local area network has multiple subnets, or you have remote users connected via a VPN.

Chapter 4: Setting Up Trunks and Lines

Specify the IP address and mask for each subnet in the following format: `nnn.nnn.nnn.nnn/mmm.mmm.mmm.mmm`, where `n` is used for the IP address and `m` is used for the mask. Separate multiple entries with a comma. For example, this entry defines two subnets:

`172.18.99.0/255.255.255.0, 192.168.55.0/255.255.255.0`

- **SIP BindPort:** This is the UDP port that SIP services will use. If required by your VoIP provider, you can set the default (5060) to another value. If you change the port value, then you will have to do the following:
 - Open this port on any router/firewall between the Hybrid PBX and VoIP Gateway and the VoIP provider.
 - If you are using phones that are not being automatically provisioned by the Hybrid PBX and VoIP Gateway, then you must configure the phones to register to `ip_address: port`, where `ip_address` is the LAN port IP address and `port` is the new SIP BindPort value.
- **RTP Ports:** This field defines the range of UDP ports used by the SIP phones. By default, the range is set to 10000–20000. You can customize this to reflect the number of users at your location. (Each call requires 2 ports: one for RTP and one for RTCP.)

Step 3: Configure port forwarding

If the connection to the Internet is behind a router or firewall, access to the VoIP lines will likely be blocked. You will generally have to open ports on the firewall and configure port forwarding on any intervening router that is doing network address translation (NAT).

By default, the following ports need to be unblocked on the firewall and forwarded by the router: UDP ports 5060 (set using SIP BindPort), plus all ports between 10000 and 20000 (set using RTP Ports). (See Step 2 for details.)

Configure the router to forward the ports to the IP address assigned to either the LAN or WAN ports on the Hybrid PBX and VoIP Gateway (depending on which one makes the connection to the router).

Verifying line status

During the installation of VoIP lines you can use the Status page to view the current status of the VoIP line. See “VoIP” in Section 2.7.1.

4.3 Editing an SIP Trunk/Line

When you edit an SIP trunk/line you have access to advanced parameters that are not available when you add the trunk/line.

1. Click “Edit” on the Trunks/Lines list page next to an existing trunk. This displays the Trunks/Lines edit page.

Trunks / Lines

Name: VoIP1

Type: VoIP

Protocol: SIP

Authentication Method: User Name

Provider:

IP Address / Domain:

User Name:

Password:

Port: 5060

Auth:

Call Limit:

From User:

Register: ☒

Qualify: yes

DTMF Mode: rfc2833

Codec 1: ulaw

Codec 2: gsm

Video Codec:

Music on Hold: DefaultMoH

Cancel Save

Figure 4-4. Trunks/lines edit page.

2. Configure the following settings using the information from your VoIP provider.

- Name: Specify a name to identify the line on the Hybrid PBX and VoIP Gateway.
- Type: Select VoIP.
- Protocol: Select the protocol supported by the line: SIP or IAX2.
- Authentication Method: Select the method that will be used to identify the Hybrid PBX and VoIP Gateway to the VoIP provider.
- Provider: (Only available if SIP is selected for Protocol.) Select your VoIP provider. The Hybrid PBX and VoIP Gateway automatically defines configuration settings a number of common providers. If your provider is not on this list, you will need to create a custom template for it. See “Creating a custom VoIP provider template” in Section 4.6.
- IP Address/Domain: Specify the IP address or domain name provided by your VoIP provider for your VoIP lines.
- User Name: (Only available if User Name is selected for Authentication Method.) Specify the user name assigned to you by your provider.

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- Password: (Only available if User Name is selected for Authentication Method.) Specify the password assigned to you by your provider.
- Port: Set the port on which the VoIP service is offered. For most installations this is port 5060.
- Auth: Select MD5 password authentication if required by your VoIP provider.
- Call Limit: Specify the number of calls that can be made simultaneously with this trunk.
- From User: Specify the user to put in the “from” field instead of the callerID number. This is not a required setting although some VoIP providers like it to be configured. Enabling this setting limits the ability to forward the callerID number of a caller when calls are forwarded to a cell phone using the follow me feature.
- Register: When enabled, the Hybrid PBX and VoIP Gateway registers with the VoIP provider with a username and password.
- Qualify: Specify how often a packet is sent to determine if your VoIP provider is still online. If the provider does not respond within two seconds to seven retries in a row, the VoIP service is marked as unreachable.
 - yes: Send one packet every 60 seconds.
 - no: Do not monitor the VoIP provider.
 - mmm: Send one packet every mmm milliseconds.
- DTMF Mode: Select the signaling protocol required by your VoIP provider.
- Codec 1: Select an audio codec supported by your VoIP provider.
- Codec 2: Select an audio codec supported by your VoIP provider.
- Video codec: Select the video codec supported by your VoIP provider.
- Music on Hold: Select the method that will be used to provide music when incoming callers on the VoIP lines are put on hold. By default the following choices are available:
 - DefaultMoH: Select the default music on hold sound file.
 - DefaultAudioIn: Select the Audio In port.

For more information, see “Setting up music on hold” in Section 6.8.

3. Click “Save” when you are done.

4. Click “Apply Configuration” and then select “System > Restart” to activate the new line/trunk.

4.4 Adding an Analog Trunk/Line

Before you can add an analog trunk/line, you need to physically connect it to an FXO port on the Hybrid PBX and VoIP Gateway as described in the Hybrid PBX and VoIP Gateway Installation Guide.

1. Click “Add” on the Trunks/Lines list page. This displays the Trunks/Lines edit page.

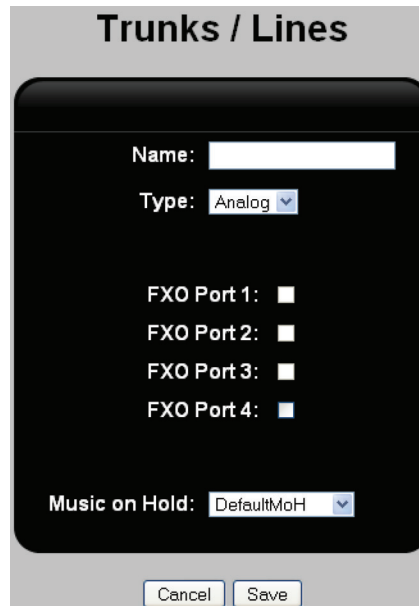


Figure 4-5. Trunks/lines edit page.

2. Configure the following settings using the information from your phone company.

- Name: Specify a name to identify the line on the Hybrid PBX and VoIP Gateway.
- Type: Select Analog.
- FXO Port 1–12: Depending on the Hybrid PBX and VoIP Gateway model, you will see 4 to 12 FXO ports listed. Enable each port that has an analog line connected to it. If you intend to split your lines into more than one line group, then only select the ports you want to include in the first group. When done configuring this line, you will need to repeat this procedure to define additional lines containing the other ports. See “Example: How to create two outgoing analog line groups” in Section 4.7.3.
- Music on Hold: Select the method that will be used to provide music when callers are on hold for incoming callers on analog lines. By default the following choices are available:
 - DefaultMoH: Select the default music on hold sound file.
 - DefaultAudioIn: Selects the Audio In port to play music from an external sound source.

For more information, see “Setting up music on hold” in Section 6.8.

3. Click “Save” when you are done. The new line will be added to the Trunks/Lines list page.

4. Click “Apply Configuration” and then select “System > Restart” to activate the new line/trunk.

4.5 Editing an Analog Trunk/Line

When you edit an analog trunk/line you have access to advanced parameters that are not available when you add the trunk/line.

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1. Click “Edit” on the Trunks/Lines list page next to an existing trunk. This displays the Trunks/Lines edit page.

The image displays two side-by-side screenshots of the 'Trunks / Lines' configuration interface. Both windows have a title bar that reads 'Trunks / Lines'. The left window is for editing a line named 'Telephone Lines'. It features a 'Name' field with 'Telephone Lines', a 'Type' dropdown set to 'Analog', a 'Pause' dropdown set to '0', and a 'Wait for Dial Tone' dropdown set to '0'. Below these are checkboxes for 'Busy Detect' (unchecked), 'Relax DTMF' (unchecked), and four 'FXO Port' checkboxes (FXO Port 1, 2, 3, and 4, all checked). At the bottom is a 'Music on Hold' dropdown set to 'DefaultMoH'. The right window is for editing a line named 'Analog 1'. It has a 'Name' field with 'Analog 1', a 'Type' dropdown set to 'Analog', a 'Pause' dropdown set to '0', and a 'Wait for Dial Tone' dropdown set to '0'. Below these are checkboxes for 'Busy Detect' (unchecked), 'Relax DTMF' (unchecked), and two 'FXO Port' checkboxes (FXO Port 1 checked, FXO Port 2 unchecked). Both windows have 'Cancel' and 'Save' buttons at the bottom.

Figure 4-6. Trunks/lines edit page.

2. Configure the following settings using the information from your phone company.

- Name: Specify a name to identify the line on the Hybrid PBX and VoIP Gateway.
- Type: Select Analog.
- Pause: Select the number of seconds to pause before dialing on this line.
- Wait for Dial Tone: Select number of seconds to wait for a dial tone to be detected before the Hybrid PBX and VoIP Gateway dials out on the selected lines. Specify 0 to disable dial tone detection.
- Busy Detect: Enable this option when the Hybrid PBX and VoIP Gateway is having difficulty detecting hangups. In some countries, there is no signal for hangup, instead hangup is identified with a busy signal. When this option is enabled, the Hybrid PBX and VoIP Gateway will try to “listen” for busy signals on the line, if your carrier sends you a busy tone on a hangup, this should detect it.
- Relax DTMF: Enable this option if callers are having difficulty making selections in IVR menus or entering extension numbers.
- FXO Port 1–12: Depending on the Hybrid PBX and VoIP Gateway model, you will see 2 to 12 FXO ports listed. Enable each port that has an analog line connected to it. If you intend to split your lines into more than one line group, then only select the analog trunks/lines having the FXO ports you want to include in the first group. When done configuring this line, you will need to repeat this procedure to define additional lines containing the other ports. See “Example: How to create two outgoing analog line groups” in Section 4.7.3.
- Music on Hold: Select the method that will be used to provide music when callers are on hold on this trunk/line. By default the following choices are available:
 - DefaultMoH: Selects the default music on hold sound file.
 - DefaultAudioIn: Selects the Audio In port to play music from an external sound source.

For more information, see “Setting up music on hold” in Section 6.8.

3. Click “Save” when you are done.
4. Click “Apply Configuration” and then select “System > Restart” to activate your changes.

4.6 Creating a Custom VoIP Provider Template

A provider template is used to define basic configuration settings for a VoIP service provider. When you add a new SIP trunk/line, you can specify a provider template to automatically define a number of settings. By default, the Hybrid PBX and VoIP Gateway defines templates for the following providers:

- babyTEL
- BroadConnect
- Dezmocom
- UnifiedVox
- Unlimited
- SIPGate
- PanOrion

If you are adding trunks/lines from another provider, you can create your own templates.

4.6.1 Viewing All Custom VoIP Provider Templates

Select “PBX > Trunks / Lines > Provider Templates.” The Provider Templates list page opens.

NOTE: The default templates supported by the Hybrid PBX and VoIP Gateway do not appear in this list.



Figure 4-7. Provider templates.

This page lists all VoIP provider templates. The following information is presented for each template.

- Name: Name that identifies the template.
- Type: SIP or IAX2.
- Action: Click the appropriate button to Edit or Delete a template. The Edit button displays the Provider Templates edit page.
- Add button: To add a new template, click Add, then see “Adding a new custom VoIP provider template,” next.

4.6.2 Adding a New Custom VoIP Provider Template

1. Click “Add” on the Provider Templates list page. This displays the Provider Templates edit page.
2. For “Name,” specify a name to identify the template on the Hybrid PBX and VoIP Gateway.
3. For Type, select SIP or IAX2. The options you see will be slightly different for each type.

The image displays two side-by-side screenshots of the 'Provider Templates' configuration window. Both windows have a dark header with the title 'Provider Templates'. The left window is configured for a SIP provider, showing fields for Name, Type (SIP), DID Identifier (Normal), IP Address / Domain, Port, Auth, Register (checked), Qualify (yes), DTMF Mode (rfc2833), Codec 1 (ulaw), Codec 2 (gsm), and Video Codec. The right window is configured for an IAX2 provider, showing fields for Name, Type (IAX2), DID Identifier (Normal), IP Address / Domain, Port (4569), Auth, Register (checked), Qualify (yes), Codec 1 (ulaw), and Codec 2 (gsm). Both windows have 'Cancel' and 'Save' buttons at the bottom.

Figure 4-8. Provider templates screens.

4. Configure the following settings:

- DID: Select the Direct Inward Dialing Number option as specified by your VoIP provider.
- IP Address / Domain: The IP address or Domain provided by the VoIP provider for the VoIP lines. For example: sip.voipprovider.com is used to establish a connection with the provider.
- Port: Set the port on which the VoIP service is offered. Generally, this is port 5060.
- Auth: Select MD5 password authentication if required by your VoIP provider.
- Register: When enabled, the Hybrid PBX and VoIP Gateway registers with the VoIP provider with a username and password.
- Qualify: Specify how often a packet is sent to determine if your VoIP provider is still online. If the provider does not respond within two seconds to seven retries in a row, the VoIP service is marked as unreachable.
 - yes: Send one packet every 60 seconds.
 - no: Do not monitor the VoIP provider.
 - mmm: Send one SIP options packet every mmm milliseconds.
- DTMF Mode: Select the signaling protocol required by your VoIP provider.
- Codec 1: Select an audio codec supported by your VoIP provider.
- Codec 2: Select an audio codec supported by your VoIP provider.
- Video codec: Select a codec to use when sending/receiving video traffic.

5. Click "Save" when you are done. The new template will be added to the "Provider Templates" list page.

6. Click "Apply Configuration" to activate the new provider template. You can now make use of this template when defining a new SIP trunk/line. See "Adding a SIP trunk/line" in Section 4.2.

4.7 Creating Outgoing Line Groups

Once all your trunks/lines have been defined, you need to create outgoing line groups. Line groups let you group multiple analog lines together under a single name. This makes it easier to define outgoing call rules. Each VoIP line must be associated with its own outgoing line group.

NOTE: An outgoing line group cannot contain both VoIP and analog lines.

You don't need to create multiple line groups. However, if you have a lot of analog lines and want to do complex call routing, having multiple groups can give you more control.

NOTE: By default, the Hybrid PBX and VoIP Gateway adds all analog lines to the outgoing line group "Analog."

4.7.1 Viewing All Outgoing Line Groups

Select "PBX > Trunks/Lines > Outgoing Line Groups". The Outgoing Line Groups list page opens.



Figure 4-9. Outgoing line groups.

This page lists all defined outgoing line groups. The following information is presented for each outgoing line group.

- Name: Name that identifies the outgoing line group.
- Action: Click the appropriate button to Edit or Delete an outgoing line group. The "Edit" button displays the Outgoing Line Groups edit page.
- Add button: To add a new group, click "Add," then see "Adding a New Outgoing Line Group" in Section 4.7.2.

4.7.2 Adding a New Outgoing Line Group

1. Click "Add" on the Outgoing Line Groups list page. This displays the Outgoing Line Groups edit page.
2. For Name, specify a name to identify the outgoing line group.
3. For Type, select VoIP or FXO/T1/E1 (analog). The configuration options you see will be slightly different for each type.

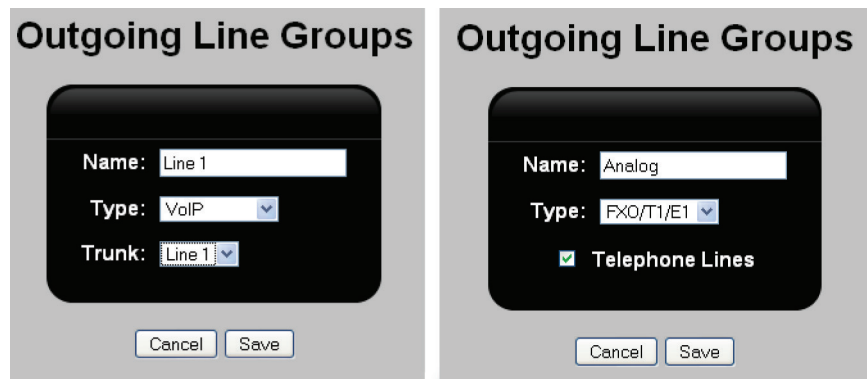


Figure 4-10. Outgoing line groups pages.

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4. Configure the following settings:

- Trunk: Select the trunk to include in the group.
- Telephone lines: This is the default analog trunk/line group to which all analog trunks/lines are automatically mapped. If you define other trunk/lines they will also appear in this list (see “Adding an analog trunk/line” in Section 4.4). Select each line that you want to include in the group.

5. Click “Save” when you are done. The new outgoing line group will be added to the Outgoing Line Groups list page.

6. Click “Apply Configuration” to activate the new outgoing line group.

4.7.3 Example: How to Create Two Outgoing Analog Line Groups

This example shows how to split four analog lines into two line groups.

1. Select “PBX > Trunks/Lines > Trunks/Lines.”

2. Add two trunks with the following settings:

The image shows two side-by-side screenshots of the 'Trunks / Lines' configuration screen. Both screens have a title bar 'Trunks / Lines' and a 'Cancel' and 'Save' button at the bottom. The left screen is for 'Analog 1/2' and the right screen is for 'Analog 3/4'. Both screens have the following settings: Name: Analog 1/2 (left) / Analog 3/4 (right), Type: Analog, Pause: 0, Wait for Dial Tone: 0, Busy Detect: [unchecked], Relax DTMF: [unchecked], FXO Port 1: [checked] (left) / [unchecked] (right), FXO Port 2: [checked] (left) / [unchecked] (right), FXO Port 3: [unchecked] (left) / [checked] (right), FXO Port 4: [unchecked] (left) / [checked] (right), and Music on Hold: DefaultMoH.

Figure 4-11. Trunks/lines settings screens.

- Set Name to Analog 1/2.
- Set Type to Analog.
- Select FXO Port 1 and FXO Port 2.
- Set Name to Analog 3/4.
- Set Type to Analog.
- Select FXO Port 3 and FXO Port 4.

3. Click “Save.”

4. Click “Apply Configuration” and then select “System > Restart.”

5. Select “PBX > Trunks / Lines > Outgoing Line Groups.”

6. Add two trunks with the following settings:

The figure shows two side-by-side screenshots of the 'Outgoing Line Groups' configuration window. Both windows have a title bar 'Outgoing Line Groups' and a dark background. The left window has 'Name: Analog Primary' and 'Type: FXO/T1/E1'. It has two checkboxes: 'Analog 1/2' (checked) and 'Analog 3/4' (unchecked). The right window has 'Name: Analog Secondary' and 'Type: FXO/T1/E1'. It has two checkboxes: 'Analog 1/2' (unchecked) and 'Analog 3/4' (checked). Both windows have 'Cancel' and 'Save' buttons at the bottom.

- Set Name to Analog Primary.
- Select Analog 1/2.
- Set Name to Analog Secondary.
- Select Analog 3/4.

Figure 4-12. Outgoing line groups.

7. Click "Save."

8. Click "Apply Configuration."

4.8 Installing 3G Support

A cellular 3G voice and data network can be used to provide a failover connection to the Internet for the Hybrid PBX and VoIP Gateway. In the event that the Hybrid PBX and VoIP Gateway detects that the normal wired Internet connection is no longer working, it can use the cellular network for voice and data connections until the wired network connection is restored. This is useful for maintaining access to VoIP trunks.

The recommended 3G device is a Rogers 3G stick—ZTE-MF 66A. Other devices may also work.

Select "System > 3G." The 3G page opens.

3G

Enable: ☐

Voice and Data: ☐

Telephone Line Failover: ☐

Ethernet Failover: ☐

Provider: Standard

DCE-DTE Options: ATQ0 V1 E1 S0=0 &C1

Packet Data Protocol: AT+CGDCONT=1,"IP",1

Dial Number: ATDT*99*1#

Probe Response Timeout: 5

Probe Interval: 10

Retry Count Till Down: 3

Retry Count Till Up: 3

Switching Delay: 60

Save

Figure 4-13. 3G page.

Configure settings as follows. Click “Save” and then “Apply Configuration” when done.

- **Enable:** Enables the 3G device, which is typically a USB thumb drive-type unit plugged into one of the available USB ports on the Hybrid PBX and VoIP Gateway. Default: not enabled
- **Voice and Data:** Enables data to be sent through the Hybrid PBX and VoIP Gateway in addition to voice. Default: not enabled
- **Telephone Line Failover:** When selected, 3G will be used as a failover if the analog phone lines fail. Default: not enabled

NOTE: VoIP carrier lines are needed to make voice calls.

- **Ethernet Failover:** When selected, 3G will be used as a failover for the Ethernet link. Default: not enabled
- **IP Address 1, Port 1, IP Address 2, Port 2:** These are the IP addresses of popular Internet services and Hybrid PBX and VoIP Gateway device ports used to verify the 3G connection. If the Hybrid PBX and VoIP Gateway does not get a response for packets sent through the specified address and ports, it will be considered a failure and the system will be unable to connect/reconnect.

Defaults:

- IP Address 1: 198.133.219.25
- Port 1: 80
- IP Address 2: 216.255.83.40
- Port 2: 80

NOTE: We recommend that you use the IP address of your ISP's DNS server as one of the test IP addresses.

- **Provider:** Select your 3G provider name from the list to automatically configure the next three fields. In the event that the 3G provider is not in the list, the settings can be tried as a starting point. Default: standard
- **DCE-DTE Options, Packet Data Protocol, Dial Number:** Set this to the value obtained from your 3G provider. However, the default values in the fields may prove to be correct. Defaults: values have been set.
- **Probe Response Timeout:** Time in seconds. If there is no response within the time specified from a packet sent as a probe, then a failure is declared. Default: 5
- **Probe Interval:** Number of seconds between probes from the Hybrid PBX and VoIP Gateway. Default: 10
- **Retry Count Till Down:** Specify how many times the Gateway will try to connect/reconnect its Ethernet connection until the connection is considered down. Default: 3
- **Retry Count Till Up:** Specify how many times the Hybrid PBX and VoIP Gateway will try to reconnect its Ethernet connection until the connection is considered active. Default: 3
- **Switching Delay:** Specify the amount of time in seconds the Hybrid PBX and VoIP Gateway will wait before switching to 3G and vice versa. Default: 60

5. Phone Provisioning

5.1 Overview

Before you can use an SIP phone to make a call through the Hybrid PBX and VoIP Gateway, it must be configured. The configuration process is called provisioning.

Provisioning can be done manually (directly on the phone), or automatically using configuration files stored on the Hybrid PBX and VoIP Gateway.

Automatic provisioning allows you to manage all configuration settings from a central location. Changes are automatically distributed to the phones when they are turned on, greatly simplifying phone management.

Manual phone provisioning may be necessary in certain cases. For example, when installing phones other than those manufactured by Aastra®, SNOM®, or Yealink®, or when configuring phones at a remote location.

5.2 Automatic Phone Provisioning

The Hybrid PBX and VoIP Gateway can automatically provision phones from Aastra, SNOM, and Yealink. This is the fastest and easiest way to deploy new phones, as you simply have to plug them into the network and they work. (If you are not using an Aastra, SNOM, or Yealink phone, automatic provisioning is still possible if you create a custom template. See “Creating a | configuration file for a non-supported phone” in Section 5.3.4.)

The Hybrid PBX and VoIP Gateway stores phone configuration settings in a template. You can create multiple templates, each one addressing a specific need. For example, all the phones in a sales group could use the settings from a Sales template, while the accounting department could use an Accounting template. Templates can even be created for individual users, such as a receptionist or CEO.

To retrieve a template from the Hybrid PBX and VoIP Gateway, a phone must first obtain the IP address of the Hybrid PBX and VoIP Gateway. The simplest way for this to occur is to install the Hybrid PBX and VoIP Gateway as the DHCP server on your LAN. This ensures that the phones will automatically retrieve their configuration settings from the Hybrid PBX and VoIP Gateway. (If another device is acting as the DHCP server, see “Configuring a third-party DHCP server” in Section 3.3.)

The sequence of events that occurs during automatic phone provisioning is as follows:

- The phone starts up.
- The phone, operating as a DHCP client the phone, obtains an IP address from the Hybrid PBX and VoIP Gateway (which is operating as a DHCP server) and the location of the boot server where it will obtain its configuration file (in this case, the Hybrid PBX and VoIP Gateway).
- The phone retrieves configuration settings from the Hybrid PBX and VoIP Gateway.
- The phone loads the settings.
- The phone is now fully configured and ready to use.

5.2.1 Setting Up the Hybrid PBX and VoIP Gateway to Automatically Provision a Phone

1. Make sure that the Hybrid PBX and VoIP Gateway is acting as the DHCP server on your network and that DHCP settings are properly configured. See “Configuring DHCP and NTP settings” in Section 3.3.
2. Configure a new template for the phone, or skip this step and use one of the pre-defined templates that come with the Hybrid PBX and VoIP Gateway. See “Configuring phone templates” in Section 5.3. If you are provisioning a non-supported phone, you must upload a custom template. See “Creating a Configuration File for a Non-Supported Phone” in Section 5.3.4.
3. Define a user for the phone as described in “Adding a new user” in Section 6.2, or skip this step and use an existing user.
4. Map the phone to the user and the template by selecting “PBX > Phones > Phone Provisioning.”

The Phone Provisioning list page opens. Initially, only the default user, called "Operator," is defined.

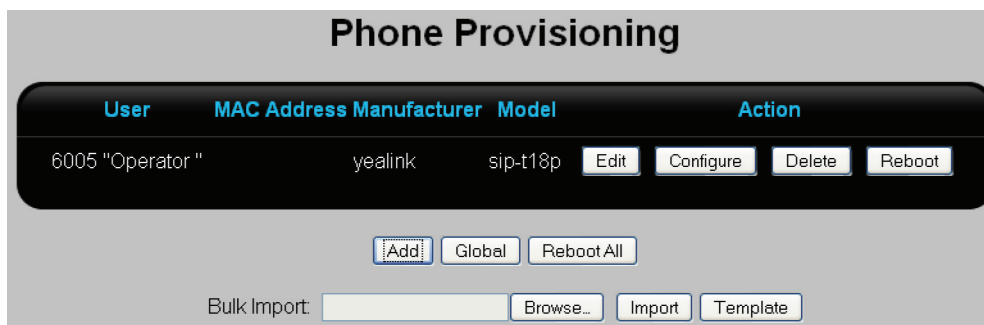


Figure 5-1. Phone provisioning page.

5. Click "Add." The Phone Provisioning edit page opens.



Figure 5-2. Phone provisioning edit page.

6. For User, select a user to associate with the phone.

7. For MAC Address, specify the MAC address of the phone. Do not enter the colons. The MAC address is usually printed on a label on the phone. It is 12 hexadecimal numbers in the format: nn:nn:nn:nn:nn:nn. For example:



Figure 5-3.

8. For Manufacturer, select the manufacturer of the phone.

9. For Model, select the model number of the phone.

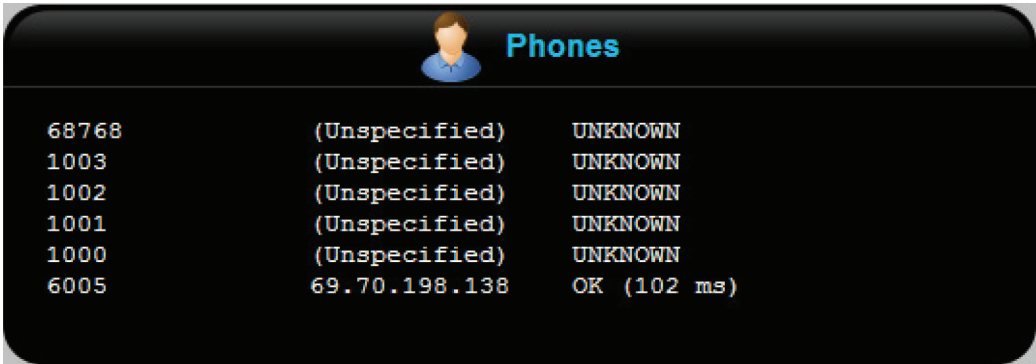
10. For Template, select the template you created in step 2 (or use a default template).

11. Click "Save."

Provisioning settings are now complete. To test your setup, do the following:

1. Plug the phone into your LAN, or directly into port 1, 2, or 3.

2. Power-on the phone. When ready, it will show the user name and extension number.
3. Check the Phones box on the Status page. The phone should appear in the list with a status of OK (xx ms). For example:



The screenshot shows a 'Phones' status page with a dark background and a light blue header. The header contains a small icon of a person and the word 'Phones' in light blue. Below the header is a table with three columns: Extension, Name, and Status. The table lists six phone extensions: 68768, 1003, 1002, 1001, 1000, and 6005. The first five extensions have a status of 'UNKNOWN', while the last one, 6005, has a status of 'OK (102 ms)'.

Extension	Name	Status
68768	(Unspecified)	UNKNOWN
1003	(Unspecified)	UNKNOWN
1002	(Unspecified)	UNKNOWN
1001	(Unspecified)	UNKNOWN
1000	(Unspecified)	UNKNOWN
6005	69.70.198.138	OK (102 ms)

Figure 5-4. Phones status page.

4. Dial a call and wait for it to connect.
5. Hang up.
6. Select "System > CDR." You should see a record for the call.

5.2.2 Automatically Provisioning Non-Supported Phones

The Hybrid PBX and VoIP Gateway contains provisioning templates for phones from Aastra, SNOM, or Yealink. If you want to take advantage of automatic provisioning for another brand of phone, you need to define a custom template for it and install it on the Hybrid PBX and VoIP Gateway. See "To create a configuration file for a nonsupported phone" in Section 5.3.4.

5.3 Configuring Phone Templates

A phone template stores the configuration settings for a specific phone model. This includes definitions for all programmable keys on the phone, as well as networking information.

The Hybrid PBX and VoIP Gateway includes predefined templates for the following phones:

- Aastra: 6730i, 6731i, 6753i, 6755i, 6757i
- SNOM: 300, 320, 360, 370, 820, 870
- Yealink: T18, T20, T22, T26, T28

You can use these templates as is, customize them, or create your own templates (for supported models only) as needed. To create a template for a non-supported phone, see "Automatically provisioning non-supported phones" in Section 5.2.2.

5.3.1 Viewing All Phone Templates

Select "PBX > Phone > Phone templates." The Phone Templates list page opens. For example, the following screen capture shows the list of default templates included with the Hybrid PBX and VoIP Gateway:

Phone Templates					
Name	Manufacturer	Model	Action		
Default_300	snom	300	Edit	Configure	Delete
Default_320	snom	320	Edit	Configure	Delete
Default_360	snom	360	Edit	Configure	Delete
Default_370	snom	370	Edit	Configure	Delete
Default_820	snom	820	Edit	Configure	Delete
Default_870	snom	870	Edit	Configure	Delete
Default_T18	yealink	sip-t18p	Edit	Configure	Delete
Default_T20	yealink	sip-t20p	Edit	Configure	Delete
Default_T22	yealink	sip-t22p	Edit	Configure	Delete
Default_T26	yealink	sip-t26p	Edit	Configure	Delete
Default_T28	yealink	sip-t28p	Edit	Configure	Delete
Default_6730i	aastra	6730i	Edit	Configure	Delete
Default_6731i	aastra	6731i	Edit	Configure	Delete
Default_6753i	aastra	6753i	Edit	Configure	Delete
Default_6755i	aastra	6755i	Edit	Configure	Delete
Default_6757i	aastra	6757i	Edit	Configure	Delete
Default_6739i	aastra	6739i	Edit	Configure	Delete
Add					

Figure 5-5. List of default templates.

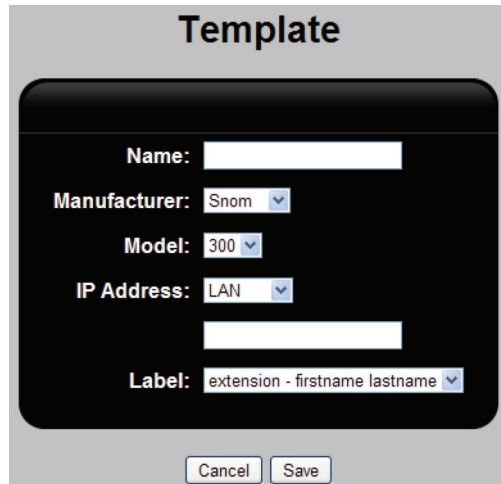
This table lists all defined templates. The following information is presented for each template.

- Name: Name that identifies the template.
- Manufacturer: Phone manufacturer.
- Model: Model number of the phone.
- Action: Click the appropriate button to Edit, Configure, or Delete a template.
- To change the general settings for an existing template, click "Edit," then see "To customize general template settings" in Section 5.3.2.
- To configure the programmable keys, click Configure, then see "To configure programmable phone keys" in Section 5.3.3.
- Add button: To add a new template, click "Add," then see "To customize general template settings" in Section 5.3.2.

NOTE: If the Configure button for a template is grayed-out, it means that the phone does not have configurable keys.

5.3.2 Customizing General Template Settings

When you click Add or Edit on the Phone Templates list page, you open the Template edit page. If you are editing an existing template, the fields will contain the currently configured settings. If you are creating a new template, it will look like this:



The screenshot shows a 'Template' configuration window. It has a dark background with white text and input fields. The fields are: 'Name' (text input), 'Manufacturer' (dropdown menu showing 'Snom'), 'Model' (dropdown menu showing '300'), 'IP Address' (dropdown menu showing 'LAN'), and 'Label' (dropdown menu showing 'extension - firstname lastname'). There are 'Cancel' and 'Save' buttons at the bottom.

Figure 5-6. Template screen.

Configure the parameters as follows. Click “Save” when you are done.

- Name: Descriptive name to identify the template.
- Manufacturer: Select the manufacturer of the phone.
- Model: Select the model of the phone.
- IP Address: Address that the phone will use to reach the Hybrid PBX and VoIP Gateway. Select one of the following options.
 - LAN: Select this option if the phone will connect to the Hybrid PBX and VoIP Gateway via its LAN port.
 - WAN: Select this option if the phone will connect to the Hybrid PBX and VoIP Gateway via its WAN port. Specify the Hybrid PBX and VoIP Gateway WAN port IP address.
 - SIP external IP: Specify the external IP address used to configure the SIP line. See “Step 2: Configure SIP/RTP support” in Section 4.2.
 - Custom: Select this option if the phone will connect to another Hybrid PBX and VoIP Gateway after being provisioned. For example, if you have branch offices, you may want to provision all phones using the Hybrid PBX and VoIP Gateway at the main office, but have the phones register and make calls through the Hybrid PBX and VoIP Gateway installed at the branch office.
- Label: Select the format in which the user’s name will display on the phone.

5.3.3 Configuring Programmable Phone Keys

When you click “Configure” on the Phone Templates page, you open the Template edit page. If you are editing an existing template, the fields will contain the currently configured settings. If you are creating a new template, it will be blank.

Each phone type will display a different set of configurable elements. The following screen captures give you an idea of the values available for each manufacturer.

The configurable settings for Type, Label, Value and other fields on these pages are defined in the documentation for your phone. Refer to this documentation for details on how these features work and how to define parameter settings.

Aastra phones

Programmable keys on Aastra phones vary by model. The following screen captures show excerpts from the default templates for some Aastra models.

Template - Aastra - Default_6730i

HK	Type	Value	Line
1	Speeddial	6050	1
2	Icom		1
3	Speeddial	"#,1,700"	1
4	Do Not Disturb		1
5	Save	SAVE	1
6	Delete	DELETE	1
7	Directory		1
8	Paging	224.168.168.168:34567	1

Figure 5-7. Aastra phone 6730i.

Template - Aastra - Default_6739i

SK	Type	Label	Value	Line
1	Call Forward	Forward		1
2	Speeddial	Voicemail	6050	1
3	Icom	Intercom		1
4	Do Not Disturb	DND		1
5	Paging	Page	224.168.168.168:34567	1

Figure 5-8. Aastra phone 6739i

Template - Aastra - Default_6757i

TK	Type	Label	Value	Line
1	Speeddial	Vmail	6050	1
2	Do Not Disturb	DND		1
3	Directory	Directory		1
4	Speeddial	ParkCall	"#,1,700"	1
5	Paging	Page	224.168.168.168:34567	1

Figure 5-9. Aastra phone 6757i

Field descriptions

- HK/SK/TK: Function key number on the phone.
 - HK = Hard key
 - SK = Softkey
 - TK = Top key. Keys across the top of the phone’s display.
- Type: Type of action to associate with the function key. For details, see the documentation from the phone’s manufacturer.
- Label: Specify the text that will appear as the label on the phone for this function key.
- Value: Parameter value for the action you selected for Type. For details, see the documentation from the phone’s manufacturer.
- Line: Line number on the phone.

SNOM phones

All SNOM phones have the same configurable value per key. The number of keys will vary according to model. For example, for model 820, you can program the following keys:

Template - Snom - Default_820

F Key	Type	Key Event	Value
P1	Line		
P2	Line		
P3	Line		
P4	DTMF		#8

Cancel

Save

Figure 5-10. SNOM phone template.

Field descriptions

- F Key: Function key number on the phone.
- Type: Type of action to associate with the function key. For details, see the documentation from the phone’s manufacturer.
- Key Event: If you select Key Event for Type, then you can select a key event. For details, see the documentation from the phone’s manufacturer.
- Value: Parameter value for the action you selected for Type. For details, see the documentation from the phone’s manufacturer.

Yealink phones

All Yealink phones have the same configurable value per key. The number of keys will vary according to model. For example, for model T28, you can program the following keys:

Template - Yealink - Default_T28				
Key	Type	Line	Value	Pickup Value
1	Speed Dial	1	6050	
2	DTMF	1	#8	
3	DTMF	1	#9	
4	Speed Dial	1	#7	
5	Speed Dial	1	*94	
6	Call Park	1	700	
7	BLF	auto	701	*7701
8	BLF	auto	702	*7702
9	BLF	auto	703	*7703
10	BLF	auto	704	*7704

Figure 5-11. Yealink phone T28.

Field descriptions

- Key: Function key number on the phone.
- Type: Type of action to associate with the function key. For details, see the documentation from the phone's manufacturer.
- Line: Line number on the phone.
- Value: Parameter value for the action you selected for Type. For details, see the documentation from the phone's manufacturer.
- Pickup Value: Set this to the BLF prefix plus the text you entered for Value. For details, see the documentation from the phone's manufacturer.

5.3.4 Creating a Configuration File for a Non-Supported Phone

The Hybrid PBX and VoIP Gateway contains provisioning templates for phones from Aastra, SNOM, or Yealink. If you want to take advantage of automatic provisioning for another brand of phone, you need to define a custom configuration file for it and install it on the Hybrid PBX and VoIP Gateway.

The following example shows how to install a template for a Polycom phone. The procedure for phones from other manufacturers is similar.

1. Consult the Auto Phone Provisioning documentation for your phone and create the required configuration files. Polycom phones require two configuration files. Other manufacturers may only require one. Name the files as follows: MACaddress.cfg and MACaddress-Phone.cfg. All letters in the MAC address must be capitalized. For this example, the two files are:
 - 0004F202734D.cfg
 - 0004F202734D-Phone.cfg
2. Select "PBX > Phones > Phone Provisioning," then click "Add." The Phone Provisioning edit page opens.

Phone Provisioning

User

MAC Address

Manufacturer

Model

Template

Action

Yealink

SIP-T18P

Default_T18

Delete

Cancel

Save

Figure 5-12. Phone provisioning edit page.

3. Configure a Polycom phone:
- For User, select a user to associate with the phone.
 - For MAC Address, specify the MAC address of the phone. Do not enter the colons.
 - For Manufacturer, select the manufacturer of the phone. This example uses Polycom.
 - For Model, select the model number of the phone. This example uses IP321.

For example:

Phone Provisioning

User

MAC Address

Manufacturer

Model

Template

Action

7002 "Polycom User"

0004F202734D

Polycom

IP 321

Delete

Yealink

SIP-T18P

Default_T18

Delete

Cancel

Save

Figure 5-13. Polycom phone.

4. Click “Save” to return to the Phone Provisioning list page.

Phone Provisioning

User

MAC Address

Manufacturer

Model

Action

7002 "Polycom User"

0004F202734D

polycom

ip-321

Edit

Configure

Delete

Reboot

Add

Global

Reboot All

Bulk Import:

Browse...

Import

Template

Figure 5-14. Phone provisioning list page.

5. Click “Configure.”



Figure 5-15. Phone provisioning page, empty.

6. Click "Browse" and select the file 0004F202734D.cfg.



Figure 5-16. Phone provisioning page, select file to upload.

7. Click "Upload." The file is added to the list.

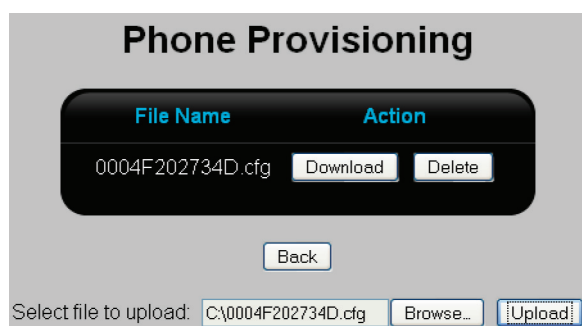


Figure 5-17. Phone provisioning page, upload file.

8. Click "Browse" and select the file 0004F202734D-Phone.cfg, then click "Upload" to add it to the list.

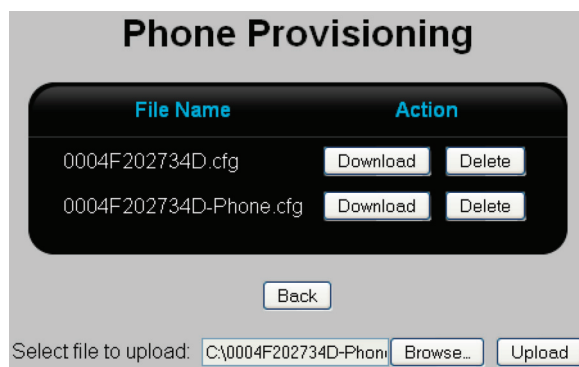


Figure 5-18. Phone provisioning page, upload another file.

9. The configuration files are now loaded and the phone can now be automatically provisioned by the Hybrid PBX and VoIP Gateway.

Testing the phone

To determine if the phone is downloading the configuration files, select "Diagnostics > Log > Web server." You should see entries that show the phone (identified by its MAC address) sending a request for the custom configuration files.

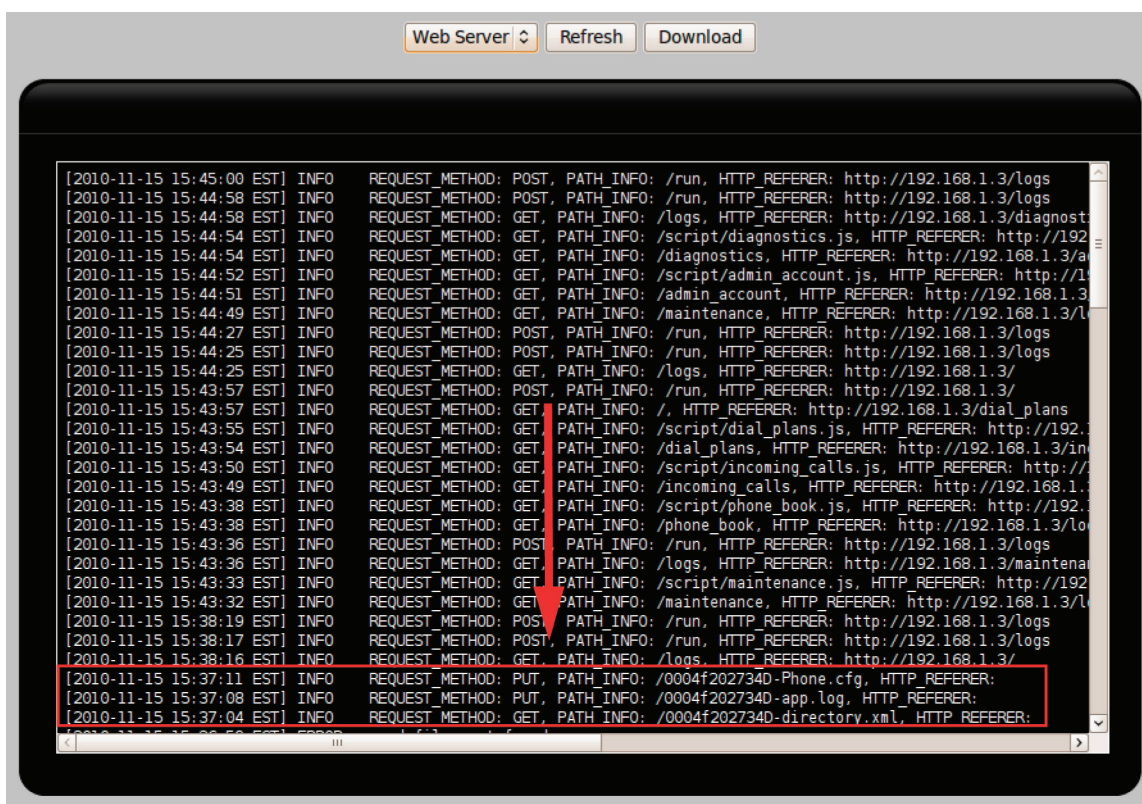


Figure 5-19. Phone sending a request for the custom configuration files.

Make sure that the phone is configured with the IP address of the Hybrid PBX and VoIP Gateway. This should occur automatically when you start the phone if it is configured as a DHCP client. If not, you will need to manually specify the Hybrid PBX and VoIP Gateway address on the phone.

5.4 Provisioning a Remote Phone

This example shows how to provision a remote phone to connect to the Hybrid PBX and VoIP Gateway via the Internet. This example uses an Aastra phone. Configuration of phones from other manufacturers will be similar.

Configuration takes place in two steps: First the Hybrid PBX and VoIP Gateway is configured and then the phone is set up at the remote location.

5.4.1 Configuring the Hybrid PBX and VoIP Gateway

1. Select "PBX > Users > Users." The Users page opens.

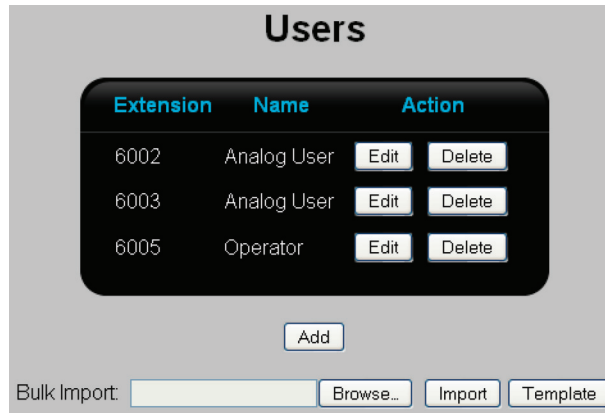


Figure 5-20. Users page.

2. Click "Add." The User page opens.

The screenshot shows the 'User' configuration page. The title 'User' is centered at the top. Below it is a form with several fields: 'Extension' (text input with '9010'), 'Type' (dropdown menu with 'SIP'), 'Template' (dropdown menu with 'PTS-external'), 'First Name' (text input with 'Remote'), 'Last Name' (text input with 'User'), 'Email Address' (text input), 'Phone Password' (text input with 'secret'), 'Cell Phone' (text input), and 'Advanced' (checkbox). The 'Advanced' checkbox is currently unchecked.

Figure 5-21. User page.

3. Define the following settings:

- Set Extension. This example uses the value 9010.
- Set Type to "SIP."
- Set Template to "PTS-external."

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- Set First Name to "Remote."
 - Set Last Name to "User."
4. Click "Save."
 5. Click "Apply Configuration."
 6. Select "PBX > Phones > Phone Provisioning." The Phone Provisioning list page opens.

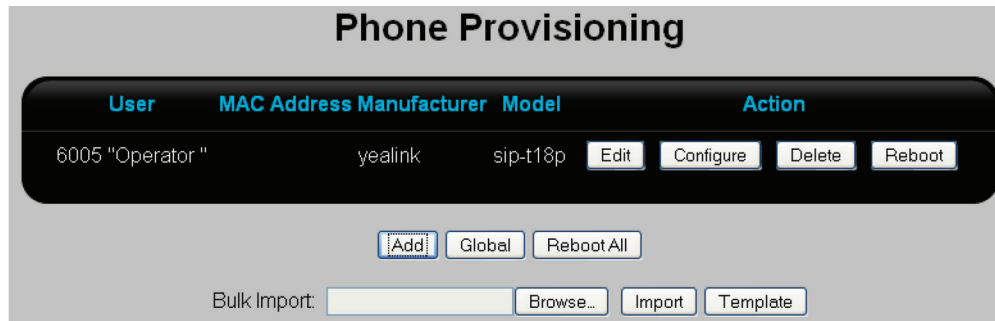


Figure 5-22. Phone provisioning list page.

7. Click "Add." The Phone Provisioning configuration page opens.



Figure 5-23. Phone provisioning configuration page.

- Set User to the new user you created.
 - Set MAC address to the MAC address printed on the phone.
 - Set Manufacturer to "Aastra."
 - Set Model to "6730i."
 - Set Template to "Default_6730i."
8. Click "Save." The Phone Provisioning list page opens with the new phone added.

Phone Provisioning

User	MAC Address	Manufacturer	Model	Action
6005 "Operator "		yealink	sip-t18p	<input type="button" value="Edit"/> <input type="button" value="Configure"/> <input type="button" value="Delete"/> <input type="button" value="Reboot"/>
9010 "Remote User"	00:AB:C9:16:C8:29	aastra	6730i	<input type="button" value="Edit"/> <input type="button" value="Configure"/> <input type="button" value="Delete"/> <input type="button" value="Reboot"/>

Bulk Import:

Figure 5-24. Phone provisioning page, new phone added.

9. Click the "Edit" button for the new phone entry. The Phone Provisioning edit page opens.

Phone Provisioning

Manufacturer:

Model:

MAC Address:

Configuration Server:

Line	IP Address	User	Label / Password
1	<input type="text" value="WAN"/>	<input type="text" value="9010 'Remote User'"/>	<input type="text" value="9010 - Remote User"/>
2	<input type="text"/>	<input type="text"/>	<input type="text"/>
3	<input type="text"/>	<input type="text"/>	<input type="text"/>
4	<input type="text"/>	<input type="text"/>	<input type="text"/>
5	<input type="text"/>	<input type="text"/>	<input type="text"/>
6	<input type="text"/>	<input type="text"/>	<input type="text"/>

Figure 5-25. Phone provisioning edit page.

10. Set Configuration Server to "WAN."

11. For Line 1:

- Set IP address to "WAN."
- For User, select an existing user.
- For Label/Password, this is displayed on phone as extension/name. If Custom is selected, this becomes the SIP password for the user on another PBX. (Enables the phone to be provisioned by the Hybrid PBX and VoIP Gateway, but register with another PBX to make calls.)

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12. Click "Save."
13. Click "Apply Configuration."

5.4.2 Configuring the Remote Phone

1. Plug the Aastra phone into the remote network.
2. Once the phone has started, press the options button.
3. Press "3." The phone should display "1. IP&MAC Addresses."
4. Press "1" to obtain the IP address of the Aastra phone.
5. Open a Web browser on a computer connected to the same network as the phone and enter http:// followed by the IP address of the Aastra phone.
6. Specify the username and password to login to the phone. The default credentials for Aastra phones are:
 - username: admin
 - password: 22222
7. Once logged in, click "Configuration Server" and configure all settings as shown.

The screenshot shows a web interface for configuring a phone's settings. On the left is a sidebar with a tree view of settings categories. A red arrow points to the 'Configuration Server' option under the 'Advanced Settings' section. The main content area is titled 'Configuration Server Settings' and contains several sections of settings:

- Settings**
 - Download Protocol: HTTP (dropdown)
 - Primary Server: 0.0.0.0 (text box)
 - Pri TFTP Path: (text box)
 - Alternate Server: 0.0.0.0 (text box)
 - Alt TFTP Path: (text box)
 - Use Alt TFTP: ☐ Enabled
 - FTP Server: (text box)
 - FTP Path: (text box)
 - FTP Username: (text box)
 - FTP Password: (text box)
 - HTTP Server: 69.70.198.140 (text box)
 - HTTP Path: pp (text box)
 - HTTP Port: 80 (text box)
 - HTTPS Server: (text box)
 - HTTPS Path: (text box)
 - HTTPS Port: 443 (text box)
- Auto-Resync**
 - Mode: None (dropdown)
 - Time (24-hour): 00:00 (dropdown)
 - Maximum Delay: 15 (text box)
 - Days: 0 (text box)
- XML Push Server List(Approved IP Addresses)**
 - (Empty text box)

At the bottom of the main content area is a 'Save Settings' button.

Figure 5-26. Configuration server settings page.

8. Click "Save Settings."
9. Restart the phone. The phone will now connect to the Hybrid PBX and VoIP Gateway and retrieve the configuration file.
10. Test by making and then receiving a call.

6. User Configuration

6.1 Viewing All Users

To view all users, select “PBX > Users > Users.” The Users page opens.

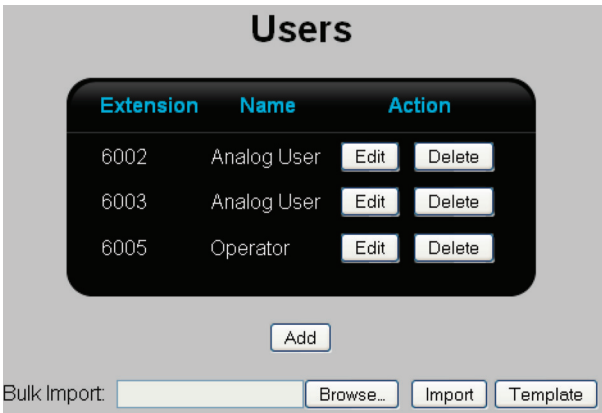


Figure 6-1. Users page.

The following information is presented for each user.

- Extension: Extension number assigned to the user. Extensions must be three to six digits long.
- Name: Name assigned to the user.
- Action: Click the appropriate button to Edit or Delete a user. The Edit button displays the User page. See “Adding a new user” in Section 6.2.

Add button

Click this button to manually add users. See “Adding a new user” in Section 6.2.

Bulk import

Use this option to add many users at once from the spreadsheet file. See “Importing user definitions from a spreadsheet” in Section 6.3.

Default users

The following users are defined by default.

- Analog User: This analog user is mapped to port FXS1 using extension 6002.
- Analog User: This analog user is mapped to port FXS2 using extension 6003.
- Operator: Sample account using extension 6005 that demonstrates configuration settings for a SIP user.

6.2 Adding a New User

NOTE: Each user should only be bound to one phone.

6.2.1 Adding a New User

Define a template for the user as described in “Working with user templates” in Section 6.4, or skip this step and use an existing template.

1. Select “PBX > Users > Users.” The Users list page opens.

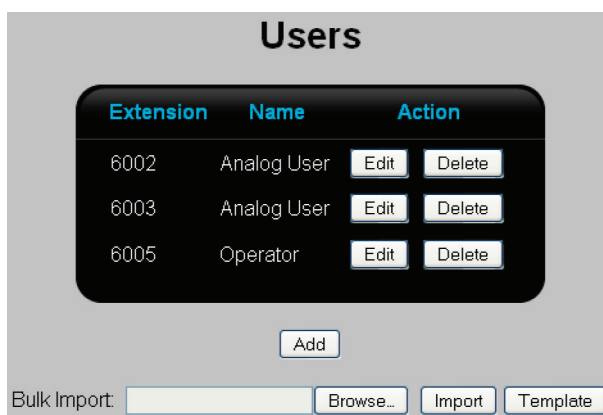


Figure 6-2. Users list page.

2. Click “Add.” The User edit page opens.

SIP

Extension:

Type:

Template:

First Name:

Last Name:

Email Address:

Phone Password:

Cell Phone:

Advanced: ☐

Analog

Extension:

Type:

FXS Port:

First Name:

Last Name:

Email Address:

Cell Phone:

Advanced: ☐

Voicemail box

Extension:

Type:

First Name:

Last Name:

Email Address:

Advanced: ☐

Figure 6-3. User edit page.

3. For Extension, specify the extension number assigned to the user. Extensions must be between three and six digits in length. You can create extensions with different lengths. For example, 234 and 2345.

4. For Type, select one of the following:

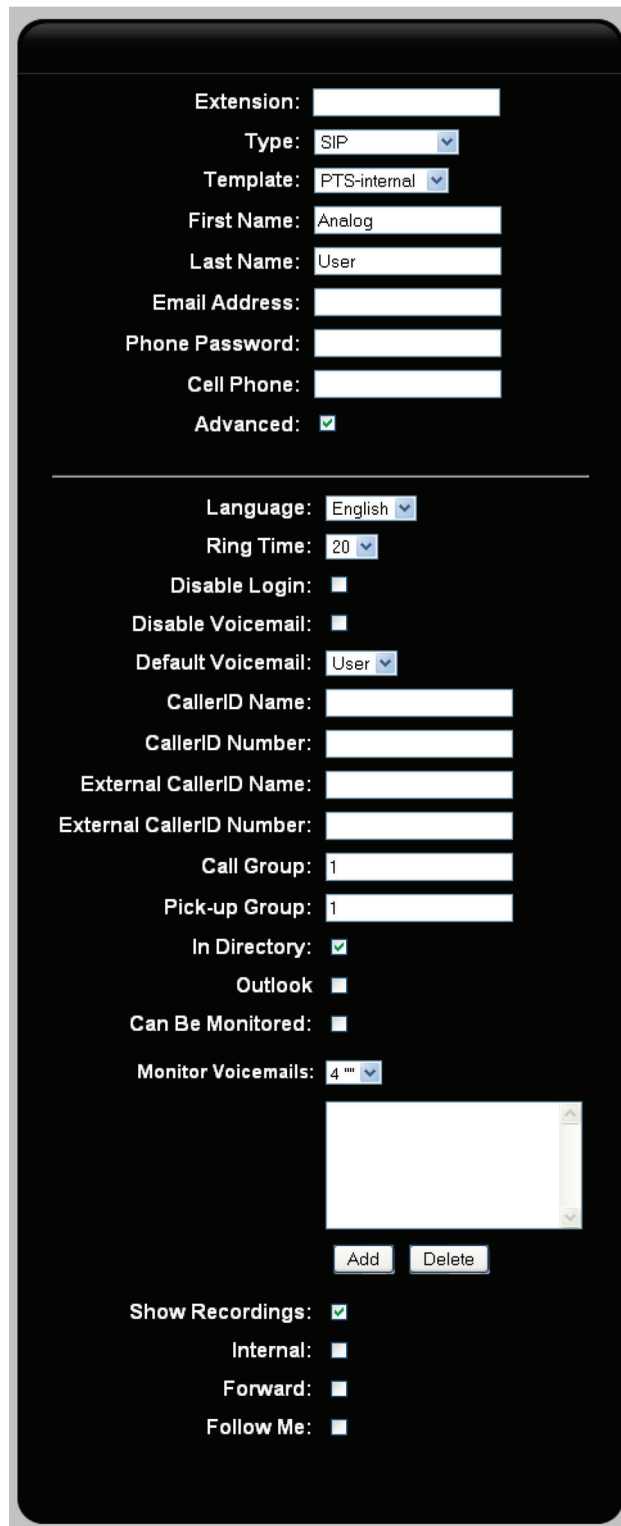
- SIP: Select this option if the user will make and receive calls on a SIP phone. See “Configuring an SIP user,” next.
- Analog: Select this option if the user will make and receive calls using an analog phone, or when adding a fax machine. See “Configuring an analog user” in Section 6.2.2.
- Voicemail Box: Select this option to set up a virtual voicemail box that is not associated with a physical user. See “Configuring a voicemail box” in Section 6.2.3.

5. Click “Save” when you are done. The new user will be added to the list on the Users page.

6. Click “Apply Configuration” to activate the new user.

6.2.2 Configuring an SIP User

Configure settings for an SIP user as follows:



The image shows a web-based configuration interface for an SIP user. The interface is divided into two main sections by a horizontal line. The top section contains fields for basic user information: Extension (text input), Type (dropdown menu set to SIP), Template (dropdown menu set to PTS-internal), First Name (text input set to Analog), Last Name (text input set to User), Email Address (text input), Phone Password (text input), Cell Phone (text input), and an Advanced checkbox which is checked. The bottom section contains more detailed settings: Language (dropdown menu set to English), Ring Time (dropdown menu set to 20), Disable Login (checkbox), Disable Voicemail (checkbox), Default Voicemail (dropdown menu set to User), CallerID Name (text input), CallerID Number (text input), External CallerID Name (text input), External CallerID Number (text input), Call Group (text input set to 1), Pick-up Group (text input set to 1), In Directory (checkbox checked), Outlook (checkbox), Can Be Monitored (checkbox), Monitor Voicemails (dropdown menu set to 4), a list box for voicemail messages with an Add button and a Delete button, Show Recordings (checkbox checked), Internal (checkbox), Forward (checkbox), and Follow Me (checkbox).

Extension:

Type: SIP

Template: PTS-internal

First Name: Analog

Last Name: User

Email Address:

Phone Password:

Cell Phone:

Advanced: ☒

Language: English

Ring Time: 20

Disable Login: ☐

Disable Voicemail: ☐

Default Voicemail: User

CallerID Name:

CallerID Number:

External CallerID Name:

External CallerID Number:

Call Group: 1

Pick-up Group: 1

In Directory: ☒

Outlook ☐

Can Be Monitored: ☐

Monitor Voicemails: 4

Show Recordings: ☒

Internal: ☐

Forward: ☐

Follow Me: ☐

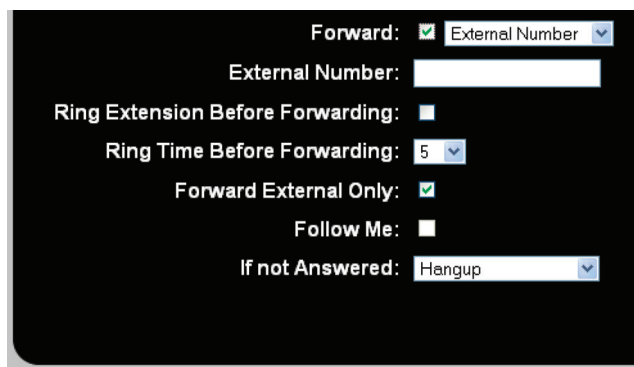
Figure 6-4. SIP user configuration screen.

- **Extension:** Specify the extension number assigned to the user. Extensions must be between three and six digits in length.
- **Type:** Select SIP which indicates that the user will make and receive calls on a SIP phone.
- **Template:** Select the template to use for all calls made by this user. To define a template, see “Working with user templates” in Section 6.4.
- **First Name:** User’s first name.
- **Last Name:** User’s last name.
- **Email address:** Specify the email address where voicemail will be sent. See “Setting up voicemail to email” in Section 8.2.
- **Phone password:** Specify the password that the SIP phone must supply when connecting to the Hybrid PBX and VoIP Gateway. If using automatic provisioning, this password is automatically configured on the phone. During bulk import, the password is automatically generated. If you are attempting to register a phone manually, then you will need to configure this password directly on the phone.
- **Cell Phone:** Specify the user’s cell phone number if available. Since the call to the cell phone is an outgoing call, add any dialing prefixes that are required by the outgoing call rule assigned to the user. For example, if the user’s cell phone number is 514-650-2232, and the outgoing call rule requires 9 to reach an outside line, you must specify: 9514-650-2232.

Other local users can call the user’s cell phone automatically by dialing 7 and the user’s extension number. For example, if the user’s extension is 2233, then any internal user can reach the cell phone by dialing 72233.
- **Advanced:** When enabled, allows configuration of the advanced settings described below.
- **Language:** Select the language in which voicemail prompts will be presented to the user. This does not set the language that callers hear when they reach the user’s voicemail box. This language is set by selecting PBX > PBX Settings > General. To install support for other languages see “Installing a language pack” in Section 10.7.
- **Ring Time:** Specify the number of seconds that the phone will ring before an incoming call is disconnected, sent to voicemail (if configured), or forwarded (if configured).
- **Disable Login:** When selected, disables access to the user portal for this user. See the Hybrid PBX and VoIP Gateway PBX User’s Guide for details.
- **Disable Voicemail:** When selected, disables the voicemail box for this user.
- **Default Voicemail:** Select the voicemail box where messages are stored by default. Select the “User” option to select the default voicemail box for this user (identified by the user’s extension). You can also assign another virtual voicemail box if available.
- **CallerID Name:** Specify the text that will appear in the callerID name field for all internal calls, or calls made to remote offices. See “Configuring callerID” in Section 6.5.
- **CallerID Number:** Specify the phone number that will appear in the callerID number field for all internal calls, or calls made to remote offices. See “Configuring callerID” in Section 6.5.
- **External CallerID Name:** Specify the text that will appear in the callerID name field for all external calls. Support for this may need to be configured by your VoIP carrier. See “Configuring callerID” in Section 6.5.
- **External CallerID Number:** Specify the phone number that will appear in the callerID number field for all external calls. Support for this may need to be configured by your VoIP carrier. See “Configuring callerID” in Section 6.5.
- **Call Group:** Specify the number of the call group to which this user belongs. See “Configuring call and pick-up groups” in Section 6.6.
- **Pick-up Group:** Specify the number of the pick-up group to which this user belongs. See “Configuring call and pick-up groups” in Section 6.6.
- **In Directory:** When enabled, the First Name and Last Name values are added to the extension directory so that they can be searched by incoming callers using the Dial by name directory feature in an IVR.

Chapter 6: User Configuration

- Outlook: When enabled, the user can make and receive calls using Microsoft Outlook. To enable Outlook support, see “Configuring network services” in Section 3.6. To configure Outlook, see the Hybrid PBX and VoIP Gateway User’s Guide.
- Can Be Monitored: When enabled, calls made by this user can be monitored. See “Monitoring calls” in Section 6.7.
- Monitor Voicemails: Select the additional voicemail boxes that the user can monitor. To view the contents of these boxes the user must login to the user portal. See the Hybrid PBX and VoIP Gateway User’s Guide.
- Show Recordings: Shows all call recordings made by the user in the user portal. To configure call recording, see “Call recordings” in Section 7.15. For information on the user portal, see the Hybrid PBX and VoIP Gateway User’s Guide.
- Internal: When enabled, this extension cannot be called from an IVR or from an incoming call on an external line. It can only be called by an internal or remote user. Applications for this include:
 - Courtesy phone: Can make, but not receive, external calls.
 - President/Director: External callers must go through the receptionist. However, internal callers can dial direct.
- Forward: When enabled, calls are forwarded if not answered. Select the destination where the call will be forwarded (extension, external number, or ring group), and then configure the following call forwarding settings:



Forward: ☒ External Number

External Number:

Ring Extension Before Forwarding: ☐

Ring Time Before Forwarding: 5

Forward External Only: ☒

Follow Me: ☐

If not Answered: Hangup

Figure 6-5. Call forwarding settings.

- External number: If you selected External number for Forward, then specify the external number where the call will be forwarded. The number will be dialed using the outgoing call rules in effect for this user.
 - Ring Extension Before Forwarding: If you selected External number for Forward, then enable this option to ring the target extension before the call is forwarded.
 - Ring Time Before Forwarding: Specify the number of seconds that the phone will ring before the call is forwarded. This replaces the value of Ring Time, which is used instead to determine how long the call will ring on the forwarded number.
- NOTE: When forwarding a call to a cell phone, a long ring time may cause the cell phone voice mail to pickup if the user does not answer. This will prevent the call from being forwarded to the users voicemail box on the Hybrid PBX and VoIP Gateway. A better solution in this case would be to use the Follow Me feature.*
- Forward External Only: When enabled, only external calls will be forwarded. Internal calls will reach the user’s extension.
 - If not Answered: Select the action to take if the call is not answered.
 - Follow me: When enabled, incoming calls are immediately placed on hold. The Hybrid PBX and VoIP Gateway then dials the user’s extension number. If the call is picked up, the user is prompted to either accept the call (press 1), or reject it (press 2). If the call is rejected, or not picked up within Ring Time, the Hybrid PBX and VoIP Gateway repeats the same operation with Phone Number 1 and Phone Number 2. If the call fails to be answered at all the numbers, then it is sent to the destination selected for If not Answered.

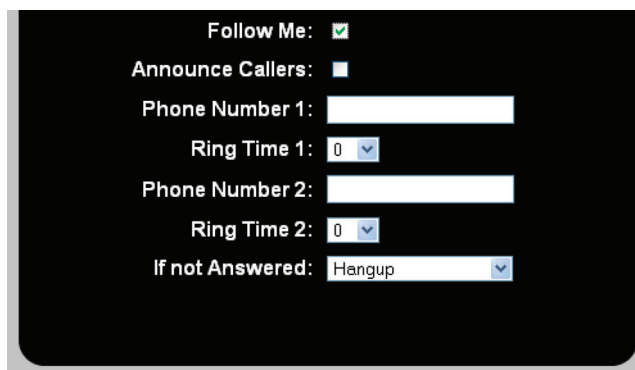


Figure 6-6. Call forwarding screen.

- **Announce Callers:** When enabled, the caller is asked to record their name. The recording is then played to the user when the call is answered as part of the prompt to accept the call.
- **Phone Number 1:** First phone number to try.
- **Ring Time 1:** Specify the number of seconds that the line will ring after dialing Phone Number 1 before attempting phone number 2.
- **Phone Number 2:** Second phone number to try.
- **Ring Time 2:** Specify the number of seconds that the line will ring after dialing Phone Number 2 before sending it to the destination configured for If not Answered.
- **If not Answered:** Select the action to take if the call is not answered.

6.2.3 Configuring an Analog User

Analog phones (or other devices), must be connected to FXS1 or FXS2. Two unique features are available on these ports:

- **Hotline:** This feature lets you create a phone that automatically dials a specific number when picked-up. When hotline is enabled, the phone is not able to receive external calls.
- **Failure support during power outage:** The G-124 and G-1212 provide fail-over support enabling an outgoing analog call to be made during a power failure. Thirty seconds after a power failure occurs, FXS2 is bridged to FXO1 on the IPBX424, and FXS2 is bridged to FXO9 on the IPXBX1224.

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Configure settings for an analog user as follows:

User

Extension: 6002

Type: Analog

FXS Port: 1

First Name: Analog

Last Name: User

Email Address:

Cell Phone:

Advanced: ☒

Hotline: ☐

Outgoing Call Rules: FXS Lines

Music on Hold: DefaultMoH

Language: English

Ring Time: 20

Disable Login: ☐

Disable Voicemail: ☐

Default Voicemail: User

CallerID Name: Analog User 1

CallerID Number:

External CallerID Name:

External CallerID Number:

In Directory: ☒

Outlook: ☒

Send Messages by Email Only: ☐

Show Recordings: ☒

Internal: ☒

Forward: ☒ 12 "test"

External Number:

Ring Extension Before Forwarding: ☐

Ring Time Before Forwarding: 5

Forward External Only: ☐

Follow Me: ☐

If not Answered: Hangup

Cancel Save Reset Voicemail Password

Figure 6-7. User screen.

- **Extension:** Specify the extension number assigned to the user. Extensions must be between three and six digits in length.
- **Type:** Select Analog which indicates that the user will make and receive calls on an analog phone.
- **FXS port:** Select the analog phone line (FXS1 or FXS2) on which this user will make calls.
- **First Name:** User's first name.
- **Last Name:** User's last name.
- **Email Address:** Specify the email address where voicemail will be sent. Voicemail messages are sent as wave file attachments. See "Setting up voicemail to email" in Section 8.2.
- **Cell Phone:** Specify the user's cell phone number if available. Since the call to the cell phone is an outgoing call, add any dialing prefixes that are required by the outgoing call rule assigned to the user. For example, if the user's cell phone number is 514-650-2232, and the outgoing call rule requires 9 to reach an outside line, you must specify: 9514-650-2232.

Other local users can call the user's cell phone automatically by dialing 7 and the user's extension number. For example, if the user's extension is 2233, then any internal user can reach the cell phone by dialing 72233.

- **Advanced:** When enabled, allows configuration of the advanced settings described below.
- **Hotline:** When enabled, this feature creates a phone that automatically dials a specified number when picked-up. If you enable the Hotline option, the following settings replace Outgoing Call Rules.

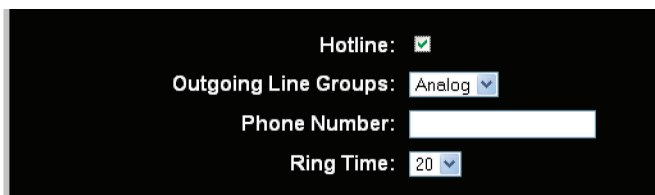
A screenshot of a configuration interface for a 'Hotline' feature. The interface is dark-themed. It shows four settings: 'Hotline:' with a checked checkbox, 'Outgoing Line Groups:' with a dropdown menu showing 'Analog', 'Phone Number:' with a text input field, and 'Ring Time:' with a dropdown menu showing '20'.

Figure 6-8. Outgoing call settings.

- **Outgoing Line Groups:** Select the line group to use to make the call.
- **Phone Number:** Specify the phone number to dial.
- **Ring Time:** Specify the number of seconds that the phone will wait for the call to be answered before terminating the call.
- **Outgoing Call Rules:** Select the rules that apply to outgoing calls made by this user.
- **Music on Hold:** Select the music to play when a caller is on hold (DefaultMoH is the default music on hold sound file), or select DefaultAudioIn (not available on the G-122), to play sound from the device connected to the Audio In port. See "Setting up music on hold" in Section 6.8.
- **Language:** Select the language in which voicemail prompts will be presented to the user. This does not set the language that callers hear when they reach the user's voicemail box. This language is set by selecting "PBX > PBX Settings > General." To install support for other languages see "Installing a language pack" in Section 10.7.
- **Ring Time:** Specify the number of seconds that the phone will ring before the call is disconnected, sent to voicemail (if configured), or forwarded (if configured).
- **Disable Login:** When selected, disables access to the user portal for this user. See the Hybrid PBX and VoIP Gateway User's Guide for details on the user portal.
- **Disable Voicemail:** When selected, disables the voicemail box for this user.
- **Default Voicemail:** Select the voicemail box where messages are stored by default. Select the User option to select the default voicemail box for this user (identified by the user's extension). You can also assign another virtual voicemail box if available.

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- **CallerID Name:** Specify the text that will appear in the callerID name field for all internal calls, or calls made to remote offices. See “Configuring callerID” in Section 6.5.
- **CallerID Number:** Specify the phone number that will appear in the callerID number field for all internal calls, or calls made to remote offices. See “Configuring callerID” in Section 6.5.
- **External CallerID Name:** Specify the text that will appear in the callerID name field for all external calls. Support for this may need to be configured by your VoIP carrier. See “Configuring callerID” in Section 6.5.
- **External CallerID Number:** Specify the phone number that will appear in the callerID number field for all external calls. Support for this may need to be configured by your VoIP carrier. See “Configuring callerID” in Section 6.5.
- **In Directory:** When enabled, the First Name and Last Name values are added to the extension directory so that they can be searched by incoming callers using the Dial by name directory feature in an IVR.
- **Outlook:** When enabled, the user can make and receive calls using Microsoft Outlook. To enable, Outlook support, see “Configuring network services” in Section 3.6. To configure Outlook see the Hybrid PBX and VoIP Gateway User’s Guide.
- **Send Messages by Email Only:** When enabled, voicemail messages will not be stored, they will only be sent by email.
- **Show Recordings:** Shows all call recordings made by the user in the user portal. To configure call recording, see “Call recordings” in Section 7.15. For information on the user portal, see the Hybrid PBX and VoIP Gateway User’s Guide.
- **Internal:** When enabled, this extension cannot be called from an IVR or from an incoming call on an external line. It can only be called by an internal or remote user. Applications for this include:
- **Courtesy phone:** Can make, but not receive, external calls.
- **President/Director:** External callers must go through receptionist. However, internal callers can dial direct.
- **Forward:** See “Forward” in Section 6.2.1.
- **Follow Me:** See “Follow Me” in Section 6.2.1.

6.2.4 Configuring a Voicemail Box

Instead of binding a user to a phone, you can bind the user to a voicemail box. This can be useful in a number of situations. For example:

- A sales department could use a voicemail box to store all incoming customer requests. All members of the sales team would have access to this box to retrieve and process messages.
- A ring group could use a voicemail box to record messages from calls that were not answered.
- A user without a SIP phone could use a voicemail box to record incoming messages and have them automatically forwarded to an email address.

Configure voicemail box settings as follows:

User

Extension:

Type: Voicemail Box ▼

First Name:

Last Name:

Email Address:

Advanced: ☒

Disable Login: ☐

In Directory: ☐

Send Messages by Email Only: ☐

Internal: ☐

Figure 6-9. Configure voicemail box settings screen.

- **Extension:** Specify the extension number assigned to the box. Extensions must be between three and six digits in length.
- **Type:** Select Voicemail Box which indicates that the user will receive calls using a voicemail box.
- **First Name:** User's first name.
- **Last Name:** User's last name.
- **Email address:** Specify the email address where voicemail will be sent. Voicemail messages are sent as wave file attachments. See "Setting up voicemail to email" in Section 8.2.
- **Advanced:** When enabled, allows configuration of the advanced settings described below.
- **Disable Login:** When selected, disables access to the user portal for this voicemail box. See the Hybrid PBX and VoIP Gateway User's Guide for details on the user portal.
- **In Directory:** When enabled, the First Name and Last Name values are added to the extension directory so that they can be searched by incoming callers using the Dial by name directory feature in an IVR.
- **Send Messages by Email Only:** When enabled, voicemail messages will not be stored, they will only be sent by email.
- **Internal:** When enabled, this extension cannot be called from an IVR or from an incoming call on an external line. It can only be called by an internal or remote user.

6.3 Importing User Definitions from a Spreadsheet

A quick way to build the user list is to import definitions from a spreadsheet as follows:

1. Select "PBX > Users > Users." The Users page opens.

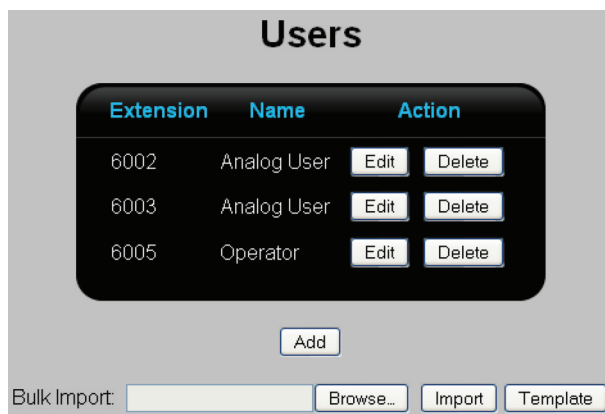


Figure 6-10. Users page.

1. Click the "Template" button to download a sample spreadsheet file to your computer.
2. Add user information to the spreadsheet according to the directions in the file. Save the file when you are done.
3. On the Users page, click "Browse" and then select the spreadsheet file on your computer. Click "OK."
4. Click "Import" to load the definitions. Each definition is checked during import. If there is a configuration error, then all definitions are discarded and no user information is imported.

NOTE: Importing overwrites all user definitions already in the system. If you want to update existing user definitions, click the "Edit" button individually for each user.

6.4 Working with User Templates

A user template enables quick configuration of a user extensions by defining a number of basic settings. In most cases, you will only need one template for your organization. However, if you have groups of users with different calling requirements it may be efficient to develop more than one template.

6.4.1 Default Templates

By default, the Hybrid PBX and VoIP Gateway contains two user templates:

- PTS_internal: Can be used when configuring internal users.
- PTS_external: Can be used when configuring external users.

6.4.2 Creating a New User Template

1. Select "PBX > Users > Templates." The Users Templates list page opens.

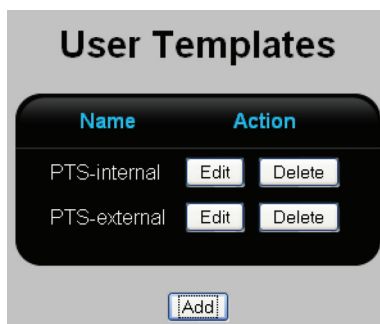
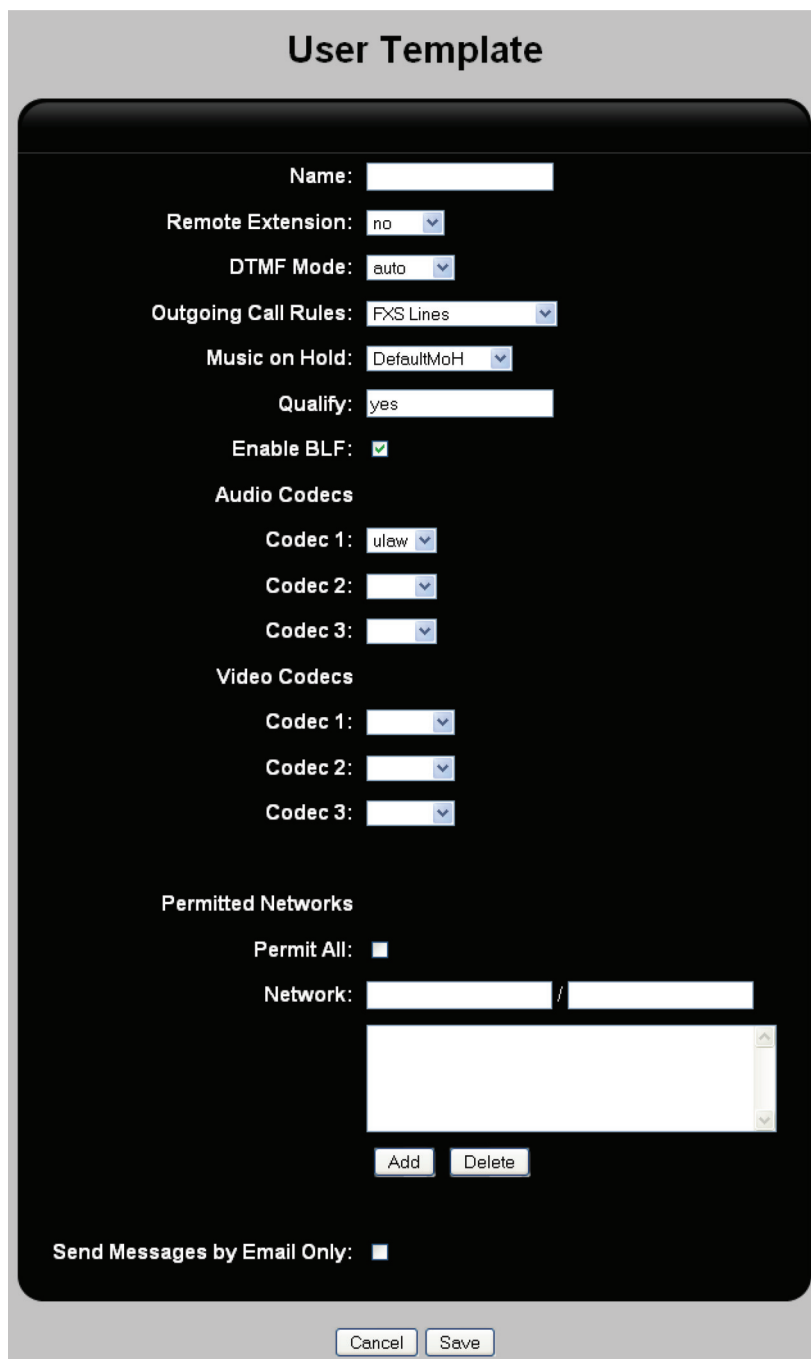


Figure 6-11. User templates list page.

2. This page shows all templates defined in the system. The following information is presented for each template.

- Name: Name assigned to identify the template.
- Action: Click the appropriate button to “Edit” or “Delete” a template. The “Edit” button displays the User Templates page.

3. Click “Add.” The User Template edit page opens.



The screenshot displays the 'User Template' configuration page. It features a dark-themed form with various settings. At the top, the title 'User Template' is centered. Below it, the form includes fields for 'Name', 'Remote Extension' (set to 'no'), 'DTMF Mode' (set to 'auto'), 'Outgoing Call Rules' (set to 'FXS Lines'), 'Music on Hold' (set to 'DefaultMoH'), and 'Qualify' (set to 'yes'). There is a checkbox for 'Enable BLF' which is checked. The 'Audio Codecs' section contains three dropdowns for 'Codec 1' (set to 'ulaw'), 'Codec 2', and 'Codec 3'. The 'Video Codecs' section also has three dropdowns for 'Codec 1', 'Codec 2', and 'Codec 3'. The 'Permitted Networks' section includes a 'Permit All' checkbox (unchecked) and a 'Network' field with a list of IP addresses below it. At the bottom of this section are 'Add' and 'Delete' buttons. A checkbox for 'Send Messages by Email Only' is also present. At the very bottom of the form are 'Cancel' and 'Save' buttons.

Figure 6-12. User template edit page.

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4. Configure settings as described below:

- Name: Specify a name to identify the template.
 - Remote Extension: Set this to “No” for internal phones or “Yes” for external phones. Use other options under advice from Black Box Technical Support.
 - DTMF Mode: Select the DTMF signalling protocol to use. DTMF mode defines the way the Hybrid PBX and VoIP Gateway sends/receives key presses to/from SIP phones.
 - Outgoing Call Rules: Select an outgoing call rule. See “Configuring outgoing call rules” in Section 7.3.
 - Music on Hold: Select the music to play when a caller is on hold (DefaultMoH is the default music on hold sound file), or select DefaultAudioIn, to play sound from the device connected to the Audio In port. See “Setting up music on hold” in Section 6.8.
 - Qualify: Determines how often a packet is sent to determine if a phone is still online. If the phone does not respond within two seconds for seven retries in a row, it is marked as unreachable.
 - yes: Send a packet every 60 seconds.
 - no: Do not monitor the phone.
 - xxx: Send a packet every xxx milliseconds.
 - Enable BLF: When enabled, the busy lamp field (BLF) can be configured allowing SIP phones to monitor the status of other phones.
 - Audio Codecs (Codec 1, Codec 2, Codec 3): Defines codecs the Hybrid PBX and VoIP Gateway will support for voice traffic. The Hybrid PBX and VoIP Gateway automatically converts the data stream if two phones on a call are using different codecs. Select the codec supported by your phone according to the following recommendations.
 - ulaw: Recommended when connecting remote phones to a G-124 or G-1212.
 - alaw: Recommended when connecting remote phones to a G-124 or G-1212.
 - g729: Recommended when connecting remote phones to a G-122.
 - gsm
 - Video Codecs (Codec 1, Codec 2, Codec 3): Defines the codecs the Hybrid PBX and VoIP Gateway will support for video traffic. The phones on both ends of a video call must use the same codec. The Hybrid PBX and VoIP Gateway does not convert video traffic between different codecs.
 - h263
 - h263p
 - h264
 - Permitted networks: Define the networks from which a SIP phone can connect to the Hybrid PBX and VoIP Gateway.
 - Permit All: When enabled, IP phones can connect to the Hybrid PBX and VoIP Gateway from any IP network.
 - Network: SIP phones can only connect to the Hybrid PBX and VoIP Gateway from the listed IP networks. To define a network, specify its IP address and network mask and click “Add.”
 - Send Message by Email Only: When enabled, voicemail is not stored. Instead, it is sent to the user via email.
5. Click “Save.” The new template will be added to the list on the User Templates page.
6. Click “Apply Configuration” to activate the new template.

6.5 Configuring callerID

CallerID settings for outgoing calls can be configured at three different places. The setting that is applied to an outgoing call is determined as follows:

1. User account. Select "PBX > Users > Users," click "Edit" for a user, then enable Advanced parameters. See "Adding a new user" in Section 6.2.1.
2. Outgoing call rule that applies to the call being made. Two settings are defined:
 - To configure the first one: Select PBX > Call Handling > Outgoing Call Rules, then click the Rules button for a rule, then click the Edit button. See "Adding an outgoing call rule" in Section 7.3.2.
 - To configure the second one: Select "PBX > Call Handling > Outgoing Call Rules," then click the "Rules" button for a rule. See "Adding an outgoing call rule" in Section 7.3.2. This is the default setting that is used if no other settings are defined.

6.6 Configuring Call and Pick-up Groups

Call and pick-up groups let you define groups of users for which calls can be answered by any user in the group from their own extension by dialing #7.

A call group defines a set of users for which calls can be answered by users belonging to a pick-up group with the same number. (Call or pick-up groups are identified by a single digit.)

If a user is part of call groups 1, 2, 3, and pick-up groups 2, 3, 4, it means that the user can pick up calls in groups 2, 3, and 4. Although the user is a member of call group 1, the user cannot pick-up calls ringing on phones in that group.

To configure the pick-up prefix, "Configuring global call handling settings" in Section 7.14.

To assign a user to a call or pick-up group, see advanced configuration settings under "Adding a new user" in Section 6.2.1.

Example

A company has three technical support teams: level 1, level 2, and level 3, and a manager that supervises them. Each team member needs to be able to pick up a call that is ringing on any phone in the team. In addition, higher level teams need to be able to pick up calls ringing in lower level teams. The manager needs to be able to pickup a call ringing in any team. To support this, call and pickup groups are defined as follows:

- Level 1 support team: Each user is assigned to call group 1 and pickup group 1.
- Level 2 support team: Each user is assigned to call group 2 and pickup group 1, 2.
- Level 3 support team: Each user is assigned to call group 3 and pickup group 1, 2, 3.
- The manager is assigned to pickup group 1, 2, 3.

6.7 Monitoring Calls

The Hybrid PBX and VoIP Gateway permits call monitoring to be done on any SIP user. To enable call monitoring:

- Configure the user's account to allow monitoring by enabling the Can Be Monitored option. See Section 6.2.1.
- Select "PBX > PBX Settings > Call Handling" and define Spy Extension and set Spy Password. See "Configuring global call handling settings" in Section 7.14.

To monitor a call:

1. Dial the Spy Extension and enter the Spy Password.
2. When prompted, dial the extension you want to monitor.

NOTE: Once the monitoring session begins, it remains open for all subsequent calls until the extension doing the monitoring hangs up.

6.8 Setting up Music on Hold

Music on hold is played any time a caller is put on hold. Music on hold can be played from a sound file stored on the Hybrid PBX and VoIP Gateway or from an audio source connected to the Audio In port.

You can define multiple music on hold profiles, enabling you to have customized music depending on how the caller was put on hold. For example, you can select different music for each of the following features:

- IVR menus (Each IVR menu can make use of many different music on hold profiles)
- Ring groups
- Conference bridges
- User templates
- Analog users
- Trunks and lines

6.8.1 Loading Pre-Recorded Music

You can play music on hold using pre-recorded sound files that you install on the Hybrid PBX and VoIP Gateway. Sound files can be any length. Sound files loop when playing, so for the best user experience try to trim the file at a point that makes looping less noticeable.

1. Record music on your computer to a sound file in gsm format. To convert a stereo wav file to gsm you can use Sox (freely available on the Internet) with the following command:

```
sox music.wav -r 8000 -c1 music.gsm resample -ql
```

For better results, try recording the wave file in 16-bit mono at a sampling rate of 8000 Hz and then run:

```
sox music.wav music.gsm
```

(Another useful program is the Switch Sound File Converter, which is also available as a free download.)

2. Select "PBX > Sound Manager > Sound Manager > Music on Hold."

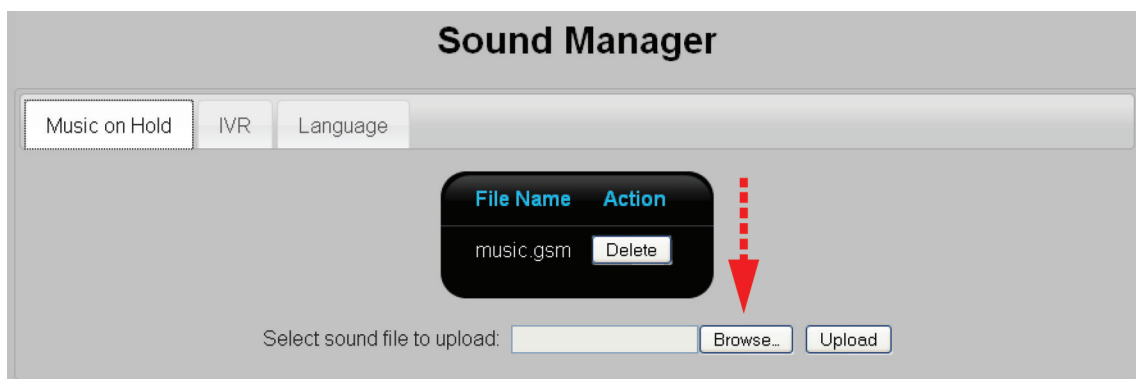


Figure 6-13. Music on Hold screen.

3. Click "Browse" at the bottom of the page, then select the sound file on your computer.
4. Click "Upload" to send it to the Hybrid PBX and VoIP Gateway.

6.8.2 Creating Music on Hold Profiles

Once you have uploaded all your sound files, you are ready to define music on hold profiles as follows:

1. Select "PBX > Sound Manager > Music on Hold." The Music on Hold list page opens.

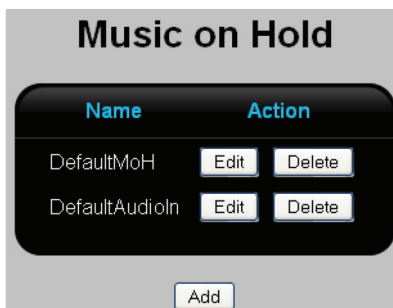
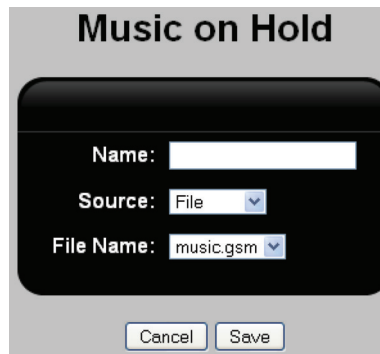


Figure 6-14. Music on Hold list page.

Initially it shows the two default profiles:

- DefaultMoH: Plays the default music on hold sound file.
 - DefaultAudioIn: Plays music from an audio source connected to the Audio In port.
2. Select "Add." The Music on Hold edit page opens. Depending on the selection you made for source, you will see different settings.



Music on Hold

Name:

Source: File ▾

File Name: music.gsm ▾

Figure 6-15. Music on Hold edit page.

3. For Name, specify a descriptive name for the sound file.
4. For Source, select "File."
5. For File Name, select the sound file you uploaded.
6. Click "Save."
7. Click "Apply Configuration."

7. Call Management

7.1 Overview

7.1.1 Handling Incoming Calls

Incoming calls can be managed using the following features:

- **Incoming Call Rules:** These rules enable you to define how incoming analog and VoIP calls will be routed to your users. Calls can be routed to a specific user, a ring group, conference room, or an IVR (interactive voice response) menu. See “Configuring incoming call rules” in Section 7.2.
- **Interactive Voice Response (IVR) system:** This feature, also known as an automated attendant, lets you build an interactive voice menu system to handle incoming calls. The Hybrid PBX and VoIP Gateway comes with a pre-defined IVR (called Sample IVR) that you can customize. See “Creating Interactive Voice Response (IVR) menus” in Section 7.5.
- **Time Frames:** This feature enables you to create rules that control how incoming calls are routed based on date and time. See “Setting up time frames” in Section 7.6.
- **Park and Page:** Lets you put a call on hold while an announcement is played on a paging system. See “Creating park and page extensions” in Section 7.7.
- **DISA (Direct Inward System Access):** Often referred to as call back or remote dial tone. This feature allows personnel who are out of the office to dial in (usually on their cell phones) and have the PBX call them back with a dial tone so that they can then dial out through the PBX. See “Configuring DISA users” in Section 7.8.
- **Ring Groups:** A ring group defines an extension number that, when dialed, rings one or more users simultaneously or in succession. See “Configuring ring groups” in Section 7.9.

7.1.2 Handling Outgoing Calls

Outgoing calls are managed using the outgoing call rules feature. Calls can be routed to outgoing lines based on the numbers user dial. See “Configuring outgoing call rules” in Section 7.3.

7.1.3 Call Accounting

The Hybrid PBX and VoIP Gateway lets you track call accounting information. Accounting codes can be created for up to three levels of detail. See “Configuring call accounting” in Section 7.4.

7.1.4 Easy Dialing of External Numbers

You can map internal extensions to automatically dial external numbers using the Phone Book feature.

This lets you create a company-wide speed dial system for frequently dialed numbers. See “Mapping external phone numbers to extensions” in Section 7.12.

7.1.5 Call Conferencing

The Hybrid PBX and VoIP Gateway provides a virtual conference room feature that allows multiple callers to meet by simply dialing an extension number. See “Creating a virtual conference room” in Section 7.11.

7.2 Configuring Incoming Call Rules

Defining incoming call rules are the first step in managing your incoming calls. You can create a set of rules for each line, or use a single rule to handle multiple lines.

Rules apply to all calls received on a line and direct all calls to a specific destination. Therefore, before defining an incoming call rule, you should create any destinations the rule will forward calls to:

- **User extension:** See “Adding a new user” in Section 6.2.
- **Ring group:** “Configuring ring groups” in Section 7.9.
- **Conference room:** “Creating a virtual conference room” in Section 7.11.

- IVR: “Creating Interactive Voice Response (IVR) menus” in Section 7.5.

7.2.1 Viewing All Incoming Call Rules

Select “PBX > Call Handling > Incoming Rules.” The Incoming Call Rules list page opens.



Figure 7-1. Incoming call rules list page.

This page lists all defined incoming call rules. The following information is presented for each rule.

- Name: Name that identifies the incoming call rule.
- Action: Click the appropriate button to Edit, Rules (configure rules), or Delete a rule. The Edit button displays the Incoming Call Rules edit page. The Rules button displays the Incoming Call Rules—Rules list page.
- Add button: To add a new rule, click Add, then see “Adding an incoming call rule” in Section 7.2.2.

7.2.2 Adding an Incoming Call Rule

Select “PBX > Call Handling > Incoming Rules.” The Incoming Call Rules list page opens.

1. Click “Add” on the Incoming Call Rules list page. The Incoming Call Rules edit page opens.

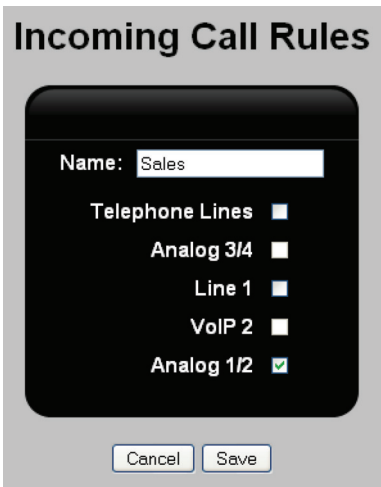


Figure 7-2. Incoming call rules edit page.

2. For Name, specify a unique name to identify the rule.

3. Underneath “Name,” select the trunks/lines that will be covered by this rule.
4. Click “Save.” The new rule will appear on the Incoming Call Rules list page.
5. Click the “Rules” button for this new rule. The Incoming Call Rules—Rules list page opens.

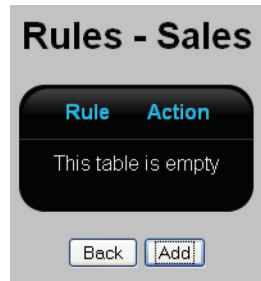


Figure 7-3. Incoming call rules—rules list page.

(This page lists all defined incoming call rules. If your VoIP line supports multiple phone numbers, you can define more than one rule. See “Example 2: Routing calls on a VoIP line with DID” in Section 7.2.4.)

6. Select “Add.” The Incoming Call Rules—Rules Edit page opens.

Figure 7-4. Incoming call rules—rules edit page.

7. Configure settings as follows:

- **DID:** (Applies to VoIP lines only) If multiple phone numbers are assigned to the VoIP line, your VoIP provider will identify each one with a direct inward dialing number (DID). Specify the DID number for which you want to route calls.
- **To:** This defines where the call is to be directed. Select from any defined user extension, IVR menu, ring group, or conference room.
- **CallerID Prefix Name:** This allows the system to prefix characters to a callerID name. This feature is useful to help a user easily identify an inbound call source. As an example, you can prefix the name “sup” to all inbound calls. When you receive a call it will have “sup” prefixed to the callerID and you will know it’s a support call.
- **CallerID Prefix Number:** This allows the system to prefix digits to the inbound callerID. This is useful, for example, when users must dial 9 and then the number to dial out. An inbound call’s callerID doesn’t have the 9 prefix. If you miss the call and press “Redial,” then the call will fail. By putting 9 in the CallerID Prefix Number field, the digit 9 will be added to the callerID.

- Answer: (Applies to Analog lines only) When enabled, it will send the correct “call answered” signal back to the telephone company even though the call has been directed to an IVR, for example (as defined in the To field).
8. Click “Save.”
9. Click “Apply Configuration.”

7.2.3 Example 1: Routing Incoming Analog Calls to Different Destinations

This example illustrates how to route incoming analog calls on two different line groups to different destinations.

- The first line group is called “Analog1.” It is composed of the physical lines attached to FXO1 and FXO2. These lines are used by the sales group. Incoming sales calls will be mapped to a ring group, with a voicemail box in case no salesperson is available to handle the call.
- The second line group is called “Analog2.” It is composed of the physical lines attached to FXO3 and FXO4. These lines are used by the support group. Incoming support calls will be mapped to an IVR that classifies the call and then sends it to the appropriate support person.

This example assumes that the sales ring group, voicemail box, and the support IVR have already been created.

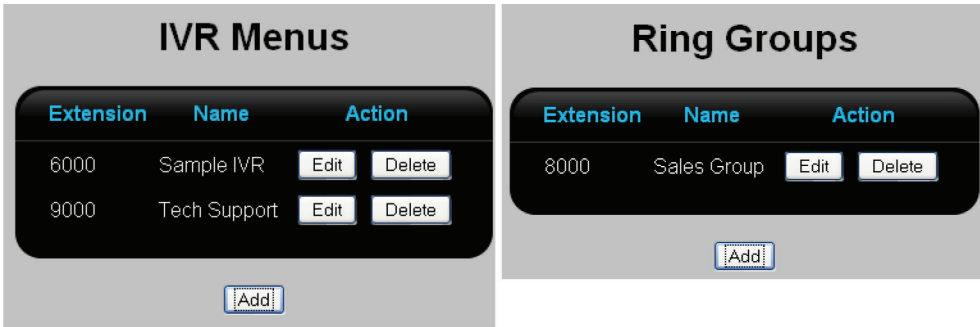


Figure 7-5. IVR menus and ring groups.

1. Select “PBX > Trunks/Lines > Trunks/Lines.” The Trunks/Lines list page opens.



Figure 7-6. Trunks/lines.

2. Click “Add.” The Trunks/Lines edit page opens.

Trunks / Lines

Name: Analog1

Type: Analog

Pause: 0

Wait for Dial Tone: 0

Busy Detect: ☐

Relax DTMF: ☐

FXO Port 1: ☒

FXO Port 2: ☒

FXO Port 3: ☐

FXO Port 4: ☐

Music on Hold: DefaultMoH

Cancel

Save

Figure 7-7. Trunks/lines edit page.

3. Set Name to "Analog1" and assign it to FXO1 and FXO2. Click "Save."

4. Click “Add.” The Trunks/Lines edit page opens.

Trunks / Lines

Name: Analog2

Type: Analog

Pause: 0

Wait for Dial Tone: 0

Busy Detect: ☐

Relax DTMF: ☐

FXO Port 1: ☐

FXO Port 2: ☐

FXO Port 3: ☒

FXO Port 4: ☒

Music on Hold: DefaultMoH

Cancel

Save

Figure 7-8. Trunks/lines edit page.

5. Set Name to “Analog2” and assign it to FXO3 and FXO4FXO2. Click Save. Both trunks are now defined.

Trunks / Lines

Name	Type	Action
Analog1	analog	<div>EditDelete</div>
Analog2	analog	<div>EditDelete</div>

Add

Figure 7-9. Assign name.

6. Select "PBX > Call Handling > Incoming Call Rules." The Incoming Call Rules list page opens.



Figure 7-10. Name/action screen.

7. Click "Add." The Incoming Call Rules edit page opens.



Figure 7-11. Incoming call rules edit page.

8. Set Name to "Sales," and select "Analog1." Click "Save."

9. Click "Add." The Incoming Call Rules edit page opens.



Figure 7-12. Incoming call rules edit page.

10. Set Name to "Support," and select "Analog2." Click "Save." Both call rules are now defined.



Figure 7-13. Both calls defined screen.

11. Click the Rules button for “Sales.” The Incoming Call Rules—Rules edit page opens.

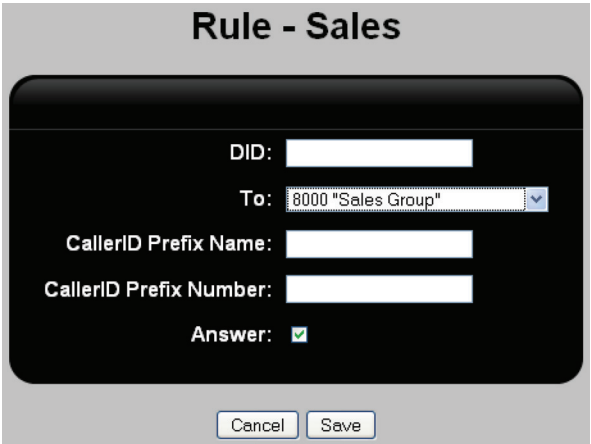


Figure 7-14. Incoming call rules—rules edit page.

12. For “To,” select 8000 “Sales Group,” which is the ring group that is defined for the sales team. Click “Save.”

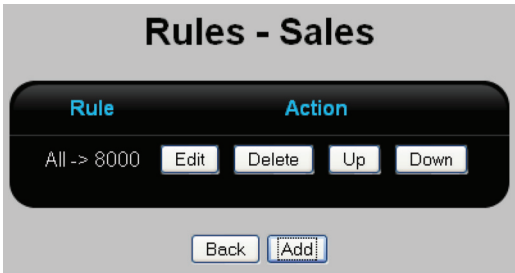


Figure 7-15. Rules—sales screen.

13. Click the “Rules” button for Support. The Incoming Call Rules—Rules edit page opens.

Figure 7-16. Rules—support screen.

14. For “To,” select 9000 “Support” which is the IVR that is defined for the support team. Click “Save.”

Figure 7-17. Rules—support screen.

15. Click “Apply Configuration.” The rules are now ready to route incoming traffic.

7.2.4 Example 2: Routing calls on a VoIP line with DID

This example shows how to direct incoming calls using Direct Inward Dial (DID) numbers. This example assumes that you have three VoIP lines and you want to route the calls as follows:

Table 7-1. Example VoIP routing calls.

DID	Route to
17247494252	Sales ring group
17247494253	Sales ring group
17247494254	Technical support IVR

This example assumes that the sales ring group, voicemail box, and the support IVR have already been created.

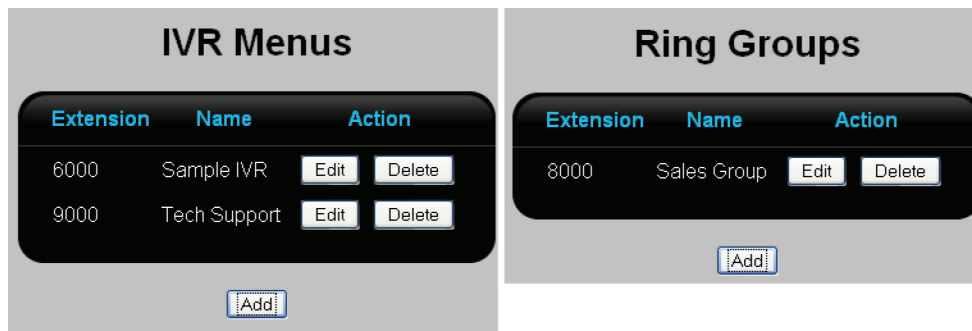


Figure 7-18. IVR menus and ring groups.

1. Select "PBX > Trunks/Lines > Trunks/Lines." The Trunks/Lines list page opens.



Figure 7-19. Trunks/lines list page.

2. Click "Add." This displays the Trunks/Lines edit page. Fill in the settings for a VoIP line as follows, then click "Save."

Trunks / Lines

Name: VoIP Line 1

Type: VoIP

Protocol: SIP

Authentication Method: User Name

Provider: babyTEL

IP Address / Domain: sip.babytel.ca

User Name: jxy3kyas

Password:

Music on Hold: DefaultMoH

Cancel

Save

Figure 7-20. Trunks/lines screen.

3. Select "PBX > Call Handling > Incoming Rules." The Incoming Call Rules list page opens.

Incoming Call Rules

Name	Action
This table is empty	

Add

Figure 7-21. Incoming call rules list page.

4. Click "Add." Give the rule a name, and select "VoIP Line 1." Click "Save."

Incoming Call Rules

Name: Sales

Telephone Lines ☐

VoIP Line 1 ☒

Cancel

Save

Figure 7-22. Naming a rule.

5. Click “Save.” The rule is added to the Incoming Call Rules list page.

Incoming Call Rules

Name	Action
Sales	<div><div>Edit</div><div>Rules</div><div>Delete</div></div>

Add

Figure 7-23. Incoming call rules list page.

6. Click “Rules.” The Incoming Call Rules—Rules list page opens.

Rules - Sales

Rule	Action
This table is empty	

Back

Add

Figure 7-24. Rules table empty.

7. Click “Add.” The Incoming Call Rules—Rules edit page opens.

Rule - Sales

DID: 7247494252

To: 7000 "Sales"

CallerID Prefix Name:

CallerID Prefix Number:

Answer: ☐

Cancel Save

Figure 7-25. Rules edit page.

- For DID, specify the number of the first line. (The length of the DID varies by VoIP provider. Some providers only use the last four digits of the phone number assigned line as the DID.)
 - For "To," select the ring group 7000 "Sales."
8. Click "Save" to return to the Incoming Call Rules—Rules list page. Click "Add," then define a rule to match the second number as follows:

Rule - Sales

DID: 7247494253

To: 7000 "Sales"

CallerID Prefix Name:

CallerID Prefix Number:

Answer: ☐

Cancel Save

Figure 7-26. Define a rule to match the second number.

9. Click “Save.” Both rules are now defined and should be visible on the Incoming Call Rules—Rules list page.



Figure 7-27. Rules—list page.

10. Click “Back.”
11. Next, a rule needs to be defined to send traffic on the third line to the support IVR. Click “Add.” Give the rule a name, and select “VoIP Line 3.”



Figure 7-28. Define a rule.

12. Click “Save.” The rule is added to the Incoming Call Rules list page.

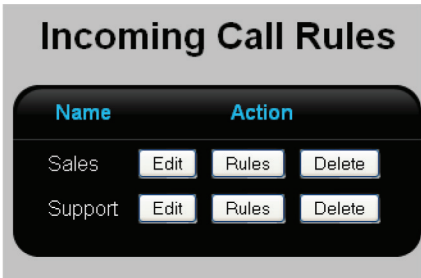


Figure 7-29. Incoming call rules page.

13. Next to Support, click “Rules.” The Incoming Call Rules—Rules list page opens.



Figure 7-30. Rules—support page.

14. Click “Add.” The Incoming Call Rules—Rules edit page opens.

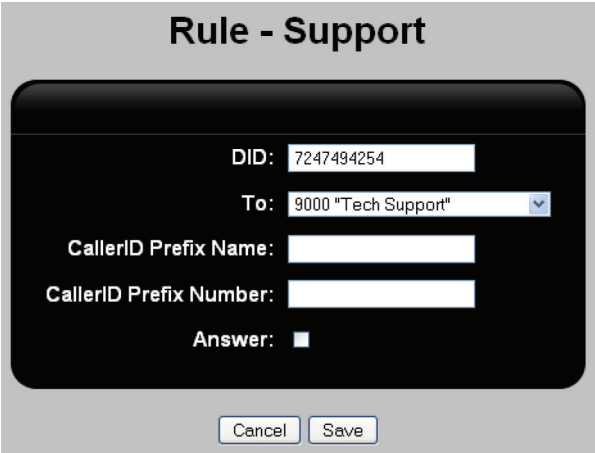


Figure 7-31. Incoming call—rules edit page.

- For DID, specify the number of the third line.
- For To, select the ring group 9000 “Tech Support.”

15. Click “Save.”

16. Click “Apply Configuration” to activate the new rules on the Hybrid PBX and VoIP Gateway.

7.3 Configuring Outgoing Call Rules

Outgoing calling rules let you control how lines are selected when an outgoing call is made.

As a simple example, you could configure an outbound rule that says all 911 calls need to use an outgoing line group defined to use analog lines, and all long-distance must use VoIP lines.

By default, there is already a default rule for the FXS ports and a basic default dialling out rule that you can modify easily. This default is there so that you can easily make outbound calls without any restrictions.

7.3.1 Viewing All Outgoing Call Rules

Select “PBX > Call Handling > Outgoing Rules.” The Outgoing Call Rules list page opens.

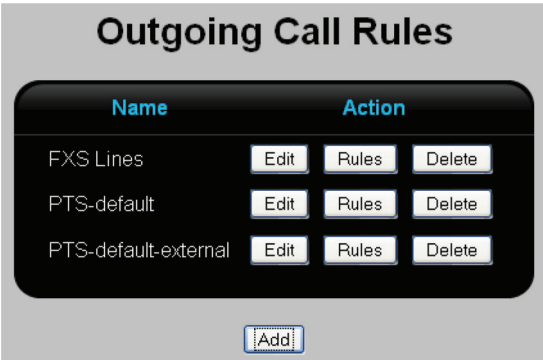


Figure 7-32. Outgoing call rules list page.

This page lists all defined outgoing call rules. The following information is presented for each rule.

- Name: Name that identifies the outgoing call rule.
- Action: Click the appropriate button to Edit, Rules (configure rules), or Delete a rule. The Edit button displays the Outgoing Call Rules edit page. The Rules button displays the Outgoing Call Rules—Rules list page.
- Add button: To add a new rule, click “Add,” then see “Adding an outgoing call rule” in Section 7.3.2.

7.3.2 Adding an Outgoing Call Rule

1. Click “Add” on the Outgoing Call Rules list page. The Outgoing Call Rules edit page opens.

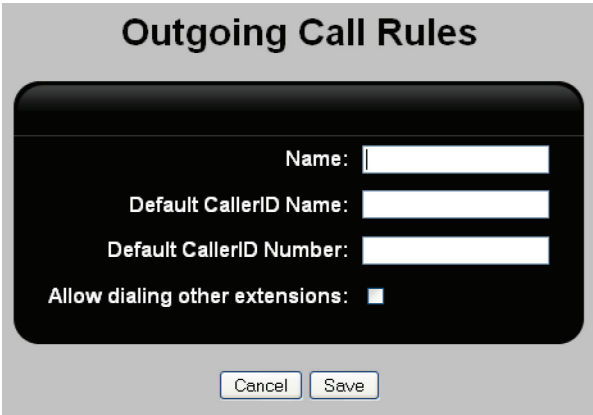


Figure 7-33. Outgoing call rules edit page.

2. Configure settings as follows:

- Name: Specify a unique name to identify the rule.
- Default CallerID Name: Specify the CallerID name that will be assigned to an outgoing call if the External CallerID Name parameter is not defined for the user or in the pattern used to make the outgoing call.
- Default CallerID Number: Specify the CallerID number that will be assigned to an outgoing call if the External CallerID Number parameter is not defined for the user or in the pattern used to make the outgoing call.
- Allow dialing other extensions: When enabled, makes calls to internal extensions exempt from these rules. In other words, allows internal extensions to be dialed freely, without restriction.

3. Click “Save.” The new rule will appear on the Outgoing Call Rules list page.

4. Click the Rules button for this new rule. The Outgoing Call Rules—Rules page opens. This displays all the patterns that make up the rule. When an outgoing call is being handled by the rule, each pattern is checked in order, from top to bottom.



Figure 7-34. Rules—FXS lines screen.

5. Select "Add." The Outgoing Call Rules—Rules Edit page opens.

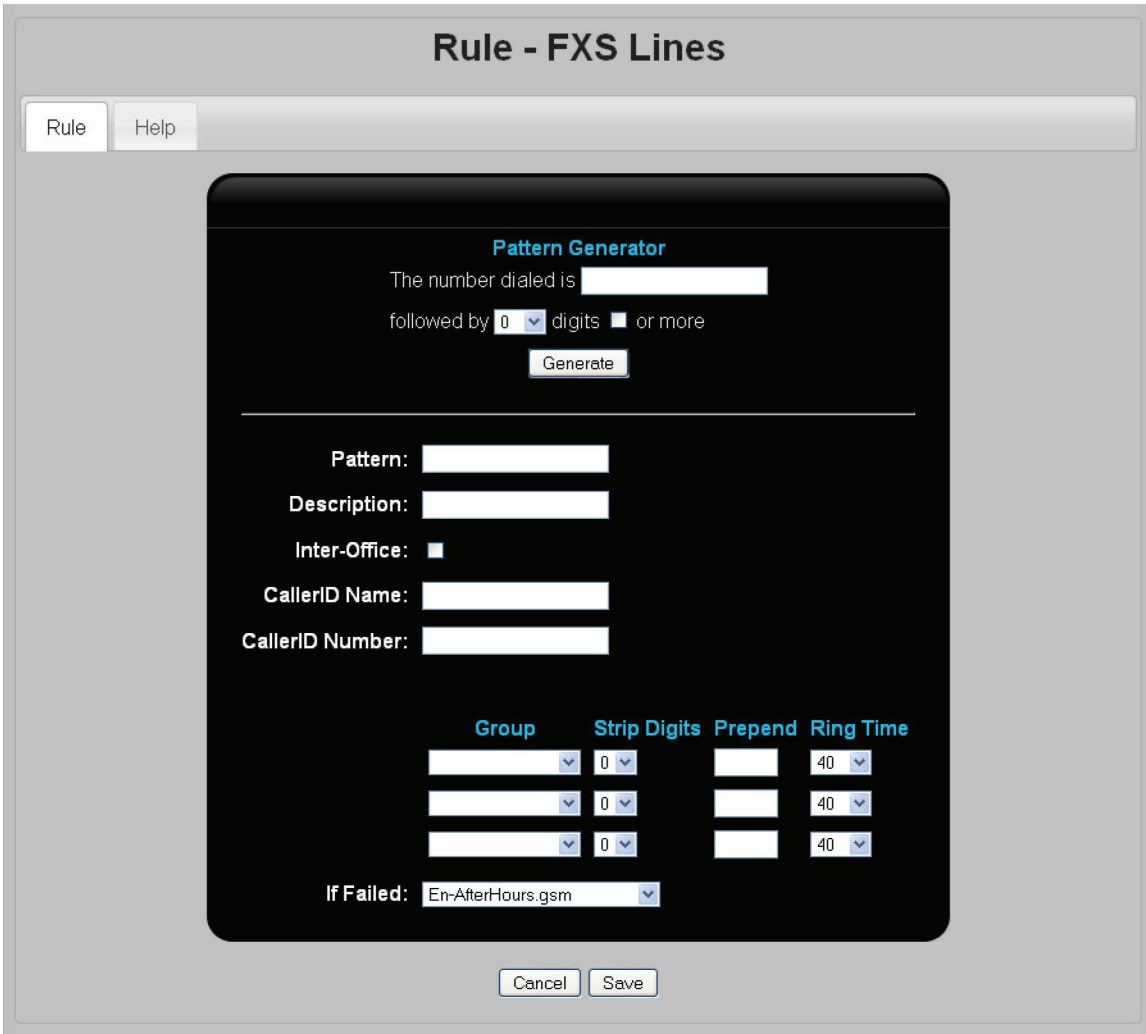


Figure 7-35. Rule—FXS lines.

6. If call accounting support is active, two additional fields are visible.

Rule - FXS Lines

Rule Help

Pattern Generator

The number dialed is

followed by digits ☐ or more

Generate

Pattern:

Description:

Inter-Office: ☐

CallerID Name:

CallerID Number:

Call Accounting:

If Failed:

Group	Strip Digits	Prepend	Ring Time
<input type="text"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text" value="40"/>
<input type="text"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text" value="40"/>
<input type="text"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text" value="40"/>

If Failed:

Cancel Save

Figure 7-36. Additional field visible when call accounting support is active.

7. Configure settings as follows:

- **Pattern Generator:** The pattern generator is a tool to help you fill in the Pattern field. See “Using the pattern generator” on the next page.
- **Pattern:** Specify the pattern that will be used to match the numbers dialed on an outgoing call. If the pattern matches the dialed number, then the settings for this pattern are applied to the dialed number, and no more patterns are checked. See “Manually creating a pattern.”
- **Description:** A description of the pattern.
- **Inter-Office:** When enabled, calls to a remote office show the user’s CallerID name and number as opposed to the external callerID of the PBX.
- **CallerID Name:** Specify the CallerID name that will be assigned to an outgoing call if the External CallerID Name parameter is not defined for the user or in the pattern used to make the outgoing call.
- **CallerID Number:** Specify the CallerID number that will be assigned to an outgoing call if the External CallerID Number parameter is not defined for the user.

- Call accounting: Select a call accounting code group. See “Configuring call accounting” in Section 7.4.
- If Failed: Select a sound recording that will be played if the call fails.
- Group: Select the outgoing line group that will be used to dial the number. Groups are dialed in the order they appear in the list (from top to bottom).
- Strip Digits: Select the number of digits you want to strip from the front of the dialed number before the call is placed.
- Prepend: Specify the digit to add to the front of the dialed number before the call is placed.
- Ring Time: Specify the number of seconds within which the call must be answered, otherwise the call is terminated.
- If Failed: Select the message to play to the user who placed the call when the call cannot be connected.

8. Click “Save.”

9. Click “Apply Configuration.”

Using the pattern generator

The pattern generator can help you to automatically build a pattern.

- The number dialed is: Specify the initial series of digits that you want to match.
- followed by x digits: Specify the number of digits (0–12) that follow the initial set.
- or more: Select this option to match any number of additional digits.

Example 1

Routing 911 calls to a specific line requires a rule that matches the three digits 911. Define it as follows:

- The number dialed is: 911
- followed by x digits: 0
- or more: Not selected.
- Resulting Pattern = 911

Example 2

Routing international calls to a specific line requires a rule that matches the prefix 011 followed by any number of digits. Define it as follows:

- The number dialed is: 011
- followed by x digits: 0
- or more: Selected.
- Resulting Pattern = _011!

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Manually creating a pattern

Use the following syntax to manually create a pattern.

Table 7-2. Manually creating a pattern parameters.

Symbol	Description
_ (underscore)	Must begin any pattern that has one or more characters in it.
X	Matches any digit.
Z	Matches any digit except 0.
N	Matches any digit except 0 or 1. Typically used to match North American area codes, which cannot begin with 1 or 0.
[n-n]	Matches any digit or letter in the brackets. For example, [01237-9] matches the numbers 1, 2, 3, 7, 8, 9.
. (period)	Matches 1 or more characters.
1	Matches 0 or more characters.

7.3.3 Example 1: Failover

In this example, calls to 911 will first try to go out on the outgoing line group Analog_Group-2, and if that is not available, then on Analog_Group-1.

The screenshot shows a 'Pattern Generator' window. At the top, it says 'The number dialed is' followed by a text box containing '911'. Below this, it says 'followed by' followed by a dropdown menu showing '0', the word 'digits', a checkbox for 'or more' (which is unchecked), and a 'Generate' button. Below a horizontal line, there are several fields: 'Pattern:' with a text box containing '911', 'Description:' with an empty text box, 'Inter-Office:' with an unchecked checkbox, 'CallerID Name:' with an empty text box, and 'CallerID Number:' with an empty text box. Below these is a table with four columns: 'Group', 'Strip Digits', 'Prepend', and 'Ring Time'. The table has three rows. The first row has 'Analog_Group-2' in the Group column, '0' in the Strip Digits column, an empty text box in the Prepend column, and '40' in the Ring Time column. The second row has 'Analog_Group-1' in the Group column, '0' in the Strip Digits column, an empty text box in the Prepend column, and '40' in the Ring Time column. The third row has an empty dropdown in the Group column, '0' in the Strip Digits column, an empty text box in the Prepend column, and '40' in the Ring Time column. Below the table is a field 'If Failed:' with a dropdown menu showing 'IVR-pls-try-again-later.gsm'. At the bottom of the window are 'Cancel' and 'Save' buttons.

Group	Strip Digits	Prepend	Ring Time
Analog_Group-2	0		40
Analog_Group-1	0		40
	0		40

Figure 7-37. Example 1: Failover screen.

- The Number dialed is field: Specify 911 because 911 identifies all emergency calls.

- Followed by: Specify 0 because there are no additional digits.
- Click the Generate button: The generated pattern is added.
- Select the line to use under Group: Analog_Group-2 is the first choice. Analog_Group-1 is used if Analog_Group-2 is not available.

7.3.4 Example 2: Stripping the First Digit

This example shows how to handle stripping of an outgoing dialing prefix. Let’s assume that users need to dial 9 to obtain an outside line.

Pattern Generator

The number dialed is

followed by digits ☒ or more

Pattern:

Description:

Inter-Office: ☐

CallerID Name:

CallerID Number:

Group	Strip Digits	Prepend	Ring Time
Analog	1	<input type="text"/>	40
<input type="text"/>	0	<input type="text"/>	40
<input type="text"/>	0	<input type="text"/>	40

If Failed:

Figure 7-38. Example 2: Stripping the first digit.

- The Number dialed is field: 9.
- Followed by: 10 or more digits (regular, phone number in North America is ten digits).
- Click the “Generate” button: This generates the pattern for you.
- Select the line to use under Group: For the first group select “Analog,” and strip 1 digit.

7.4 Configuring Call Accounting

The call accounting feature enables you to easily track calls and associate each call with a specific project or cost center. For example, you could assign all long distance calls from the sales group and support groups to their own specific accounting codes.

Once call accounting has been configured and activated, users making an outgoing call are prompted to identify how the call will be tracked. For example, they might select the department they work for from a list, and then a specific project. The Hybrid PBX and VoIP Gateway can track codes up to three levels deep.

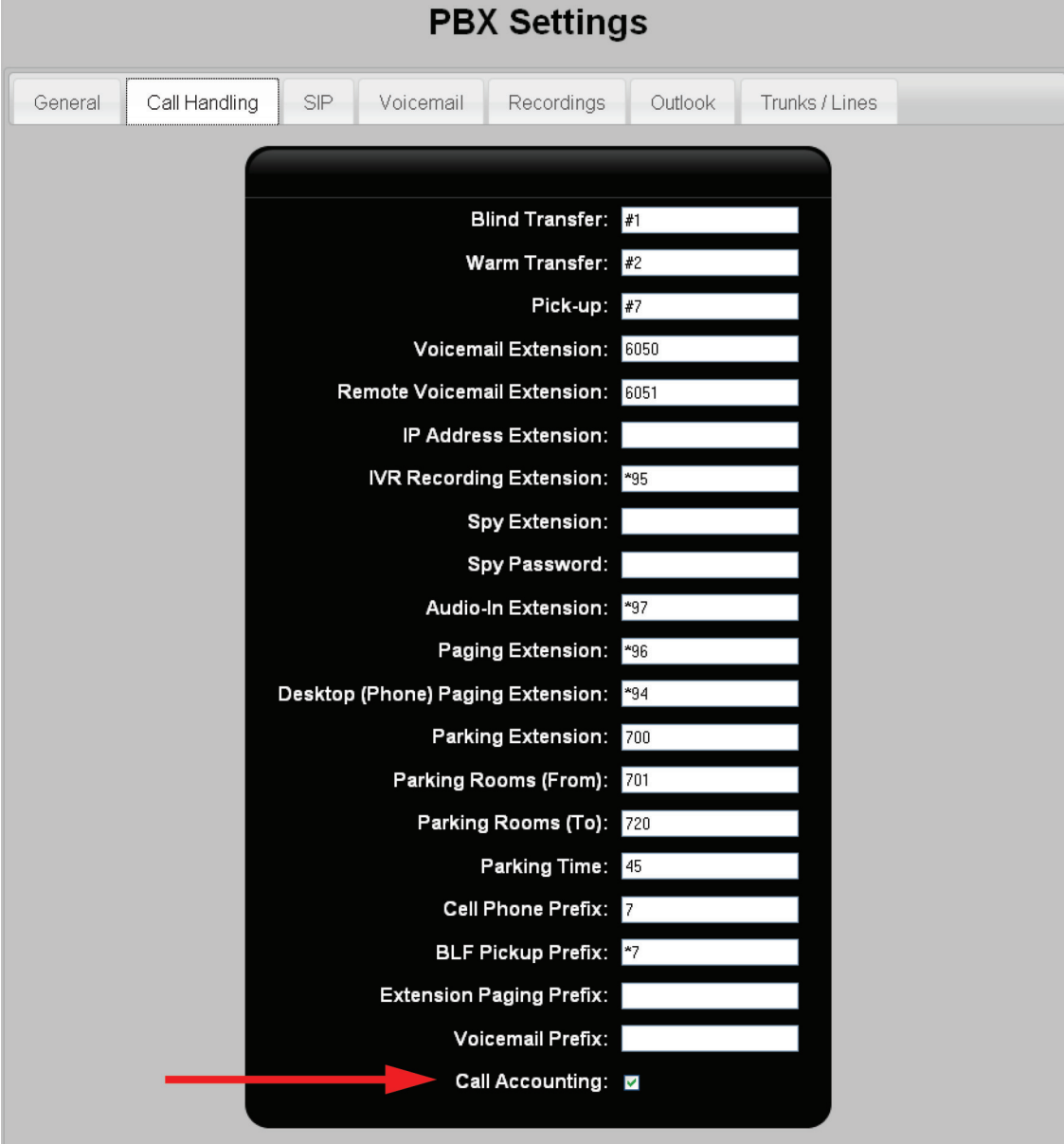
All accounting information is stored in the call detail record (CDR) which can be downloaded for processing.

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NOTE: There is no validation on codes when a user dials out. If an invalid code is entered, then a recording is played and the call is dropped.

7.4.1 Enabling Support for Call Accounting

1. Select "PBX > PBX Settings > Call Handling."



The screenshot shows the 'PBX Settings' interface with the 'Call Handling' tab selected. The 'Call Accounting' checkbox is checked and highlighted with a red arrow.

Setting	Value
Blind Transfer:	#1
Warm Transfer:	#2
Pick-up:	#7
Voicemail Extension:	6050
Remote Voicemail Extension:	6051
IP Address Extension:	
IVR Recording Extension:	*95
Spy Extension:	
Spy Password:	
Audio-In Extension:	*97
Paging Extension:	*96
Desktop (Phone) Paging Extension:	*94
Parking Extension:	700
Parking Rooms (From):	701
Parking Rooms (To):	720
Parking Time:	45
Cell Phone Prefix:	7
BLF Pickup Prefix:	*7
Extension Paging Prefix:	
Voicemail Prefix:	
Call Accounting:	<input checked="" type="checkbox"/>

Figure 7-39. Call handling screen.

2. Enable the Call Accounting checkbox.
3. Click "Save."

7.4.2 Viewing All Accounting Codes

Select "PBX > Call Handling > Call Accounting." The Call Accounting list page opens.

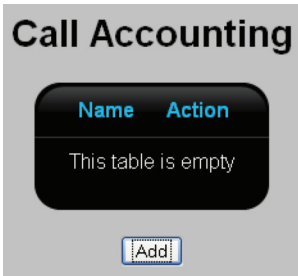


Figure 7-40. Call accounting screen.

This page lists all defined accounting code groups. Initially it is empty. The following information is presented for each group.

- Name: Name that identifies the group.
- Action: Click the appropriate button to Edit or Delete a group. The Edit button displays the Call Accounting edit page. See "Defining accounting codes," next.
- Add button: To add a new code group, click Add, then see "Define accounting codes," next.

7.4.3 Defining Accounting Codes

Before you define accounting codes, you need to define a voice recording that will be used to prompt a user for an accounting code before they make a call. See "Recording voice prompts" in Section 7.10.

1. Click "Add" on the Call Accounting list page. This displays the Call Accounting edit page.

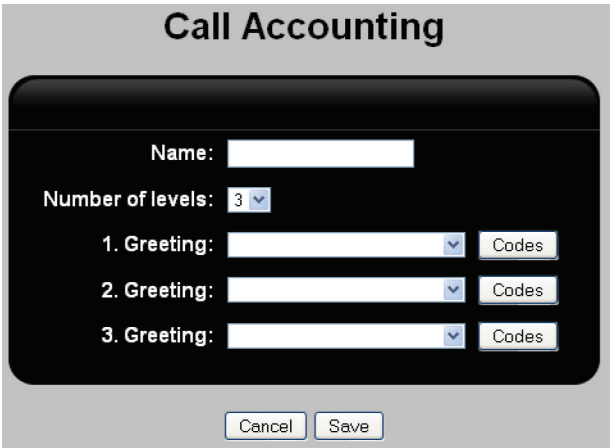


Figure 7-41. Call accounting edit page.

2. For Name, specify a name to identify the accounting group.
3. For Number of levels, choose how many levels of accounting you want to use.
4. For each level:
 - Select a Greeting from the available sound recordings. The purpose of the greeting is to prompt the caller to enter the correct accounting information for the call they are about to make.
 - Click "Codes" to open the Codes list page. This is where you define all the accounting codes that will be available at this level.



Figure 7-42. Codes list page.

5. To add a code, click “Add.” This opens the Codes edit page. This is where you define a specific code and its description.

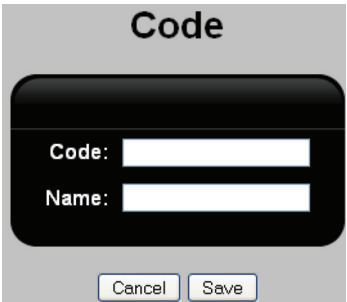


Figure 7-43. Codes edit page.

Define the following settings, then click “Save.”

- For Code, specify a numeric accounting code.
- For Name, specify a unique name to identify this code.

6. Click “Back” to return to the Call Accounting list. Define codes for the other levels as required.

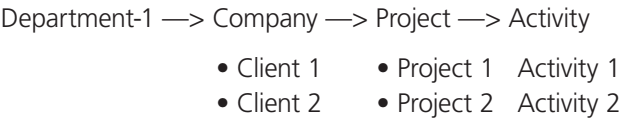
7. Click Apply Configuration when done.

Defining all three levels of accounting codes

This section uses a simple example to illustrate how to define accounting codes at all three supported levels. For this example, the levels are defined for each department in a company. This leads to the following hierarchy:

- Department: A separate set of codes is defined for each department in the company. This example shows the codes for Department-1.
- Company: A list of codes that define all client companies for Department-1. The example uses: Client1 and Client2.
- Project: A list of codes for all projects assigned to Department-1. The example uses: Project1 and Project2.
- Activity: A list of codes for all project activities for Department-1. The example uses: Activity1 and Activity2.

The hierarchy looks like this:



The following screen captures show how the hierarchy is defined using the call accounting pages.

Select “PBX > Call Handling > Call Accounting.”

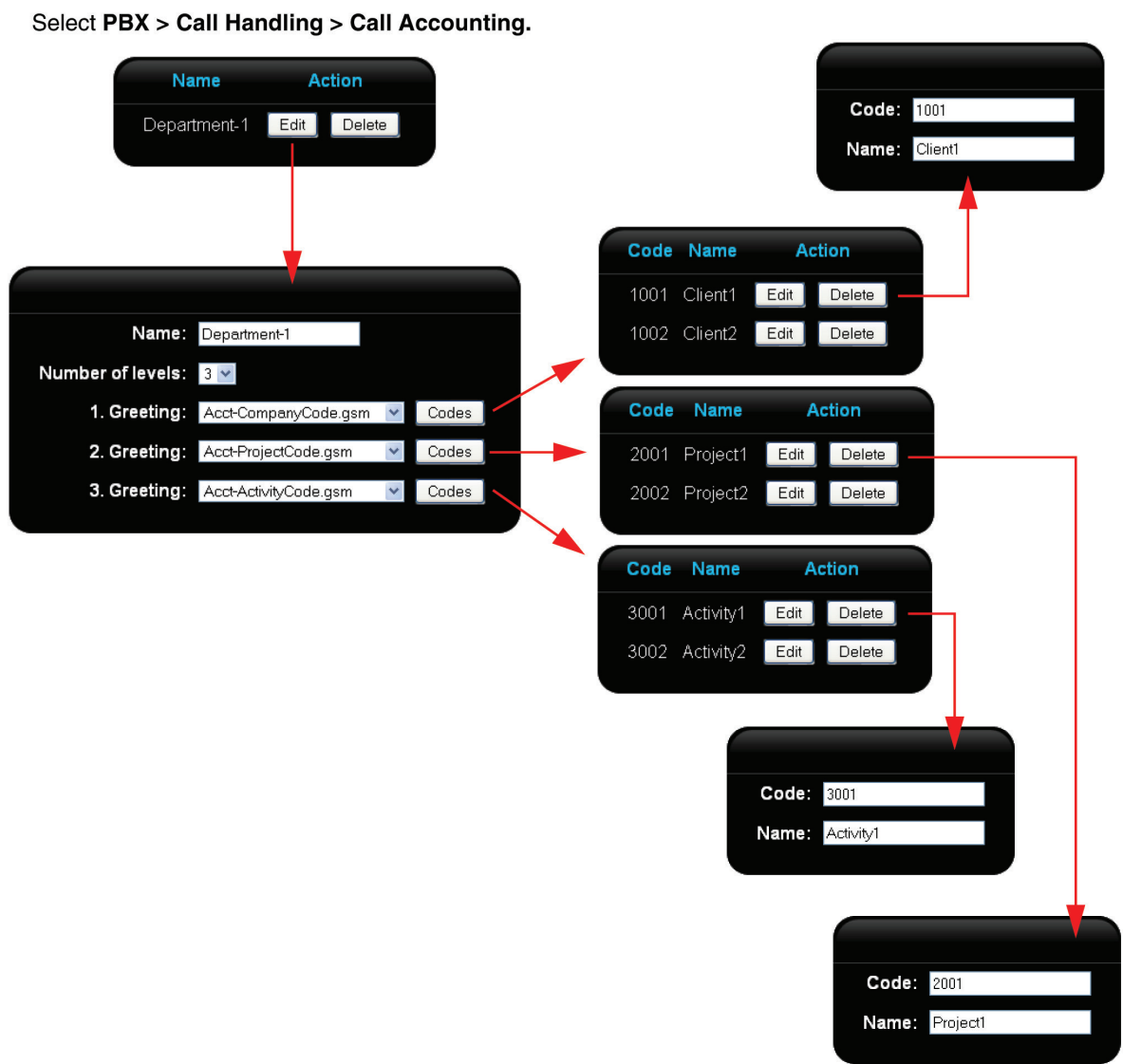


Figure 7-44. Call accounting pages hierarchy.

7.4.4 Configuring Outgoing Call Rules to Support Accounting

Once you have successfully created call accounting codes, you need to configure them in an outgoing call rule pattern so they are applied when calls are made.

- 1. Select “PBX > Call Handling > Outgoing Call Rules.” The Outgoing Call Rules list page opens.

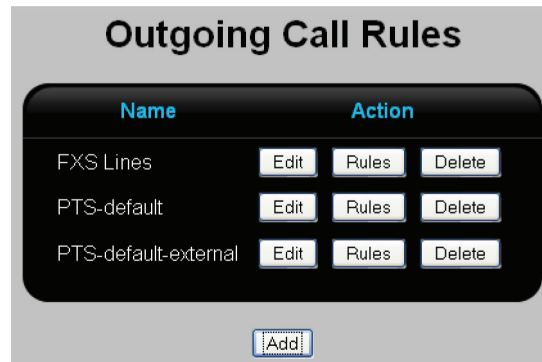


Figure 7-45. Outgoing call rules list page.

2. Click "Rules" beside the type of lines for which the accounting will be activated. For example, click "Rules" next to FXS Lines. The Outgoing Call Rules edit page opens.
3. Click "Add" to create a new rule. The Outgoing Call Rules edit page opens (See "Adding an outgoing call rule" in Section 7.3.2). Two new fields are added when accounting support is enabled.
 - Call accounting: Select a call accounting code group.
 - If Failed: Select the sound recording that will play if the call fails.

Rule - FXS Lines

Rule

Help

Pattern Generator

The number dialed is

followed by digits ☐ or more

Generate

Pattern:

Description:

Inter-Office: ☐

CallerID Name:

CallerID Number:

Call Accounting:

If Failed:

Group	Strip Digits	Prepend	Ring Time
<input type="text"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text" value="40"/>
<input type="text"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text" value="40"/>
<input type="text"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text" value="40"/>

If Failed:

Cancel

Save

Figure 7-46. Rules—FXS lines page.

- 4. Click "Save."
- 5. Click "Apply Configuration."

7.4.5 Example: Applying Accounting to All Outgoing Calls

This example enables accounting for all outgoing calls.

- 1. Select "PBX > Call Handling > Outgoing Call Rules." The Outgoing Call Rules list page opens.



Figure 7-47. Outgoing call rules page.

- 2. Click "Rules" beside the type of lines for which the accounting will be activated. For this example, click "Rules" next to FXS Lines. The Outgoing Call Rules edit page opens.

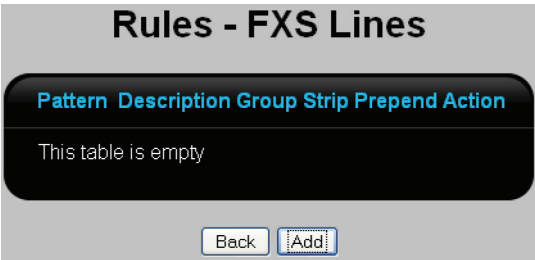


Figure 7-48. Outgoing call rules edit page.

- 3. Click "Add" to add a new rule pattern, or "Edit" to change an existing pattern.

Rule - FXS Lines

Rule Help

Pattern Generator

The number dialed is

followed by digits ☒ or more

Pattern:

Description:

Inter-Office: ☐

CallerID Name:

CallerID Number:

Call Accounting:

If Failed:

Group	Strip Digits	Prepend	Ring Time
<input type="text"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text" value="40"/>
<input type="text"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text" value="40"/>
<input type="text"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text" value="40"/>

If Failed:

Figure 7-49. Add or edit rules.

4. Leave "The number dialed is" blank.
5. For "followed by," select "6" and enable the "or more" option. This skips all internal calls that use three- to six-digit extension codes.
6. Click "Generate" to create the Pattern.
7. For Call Accounting, select "Department-1." This selects the accounting code group to use. To see how this group was defined, go to "Defining all three levels of accounting codes."
8. For "If Failed," select TryAgainLater.gsm. This is a voice prompt that tells the user that the call could not be completed, and to retry later. To create a voice prompt, see "Recording voice prompts" in Section 7.10.

7.4.6 Viewing Accounting Codes

The Hybrid PBX and VoIP Gateway tracks all accounting information on the CDR (call detail records) page. Select “System > CDR.” For example:

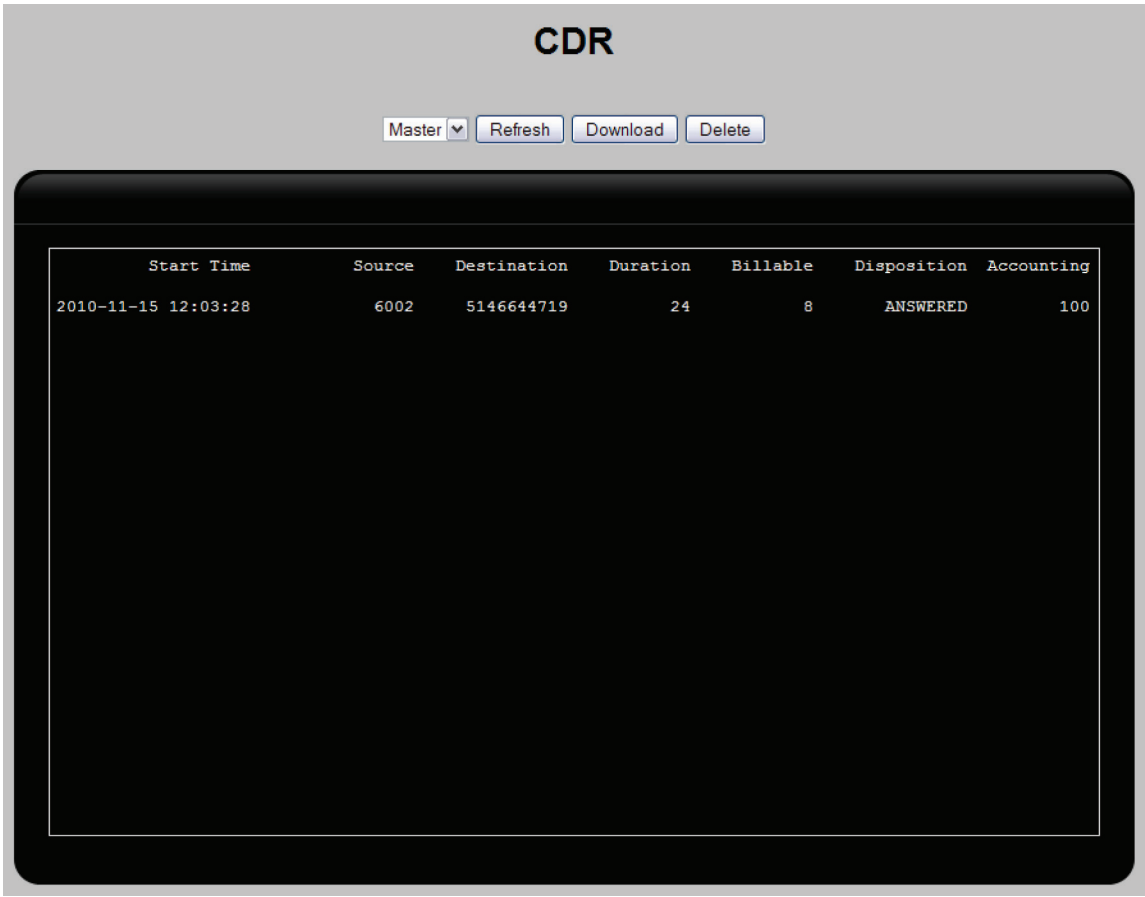


Figure 7-50. CDR screen.

Click the “Download” button to retrieve the data for processing in other applications. For more information, see “Tracking call information” in Section 7.13.

7.5 Creating Interactive Voice Response (IVR) Menus

The IVR system enables you to create an interactive voice-based menu system that can be used to route calls and provide information.

You can create a single IVR to handle all your calls, or create several IVRs, each with a different purpose. IVRs can call each other, allowing for sophisticated call processing to be done.

7.5.1 Before You Begin

- Define any extensions to which you want to route calls.
- Record all prompts (sound files) that will be played for callers. See “Recording voice prompts” in Section 7.10.

7.5.2 Creating an IVR Menu

1. Select PBX > Incoming Calls > IVR Menus. The IVR Menus page opens.

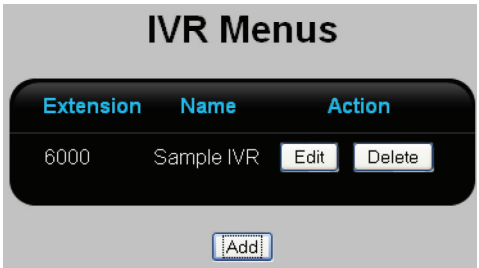


Figure 7-51. IVR menu screen.

This page shows all IVR menus defined in the system. The following information is displayed for each menu.

- Extension: Extension number that activates the menu.
 - Name: Name of the menu.
 - Action: Click the appropriate button to “Edit” or “Delete” a menu. The “Edit” button displays the IVR Menu page which allows you to edit an existing menu.
2. Click “Add.” The IVR Menu page opens.

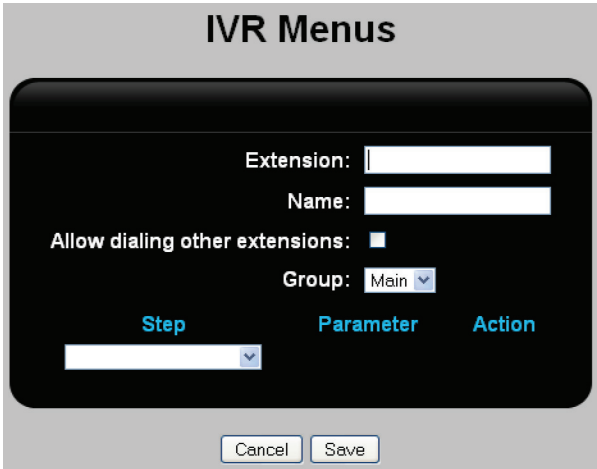


Figure 7-52. IVR menu page.

3. For Extension, specify the extension number that will be used to activate the IVR. Extension numbers must be between three and six digits in length. If you prefix the extension number with a single character, the IVR cannot be started from a phone, instead it can only be called from another IVR.
4. For Name, specify a name to identify the IVR.
5. Enable “Allow other dialing extensions” to allow users to dial extensions that are not defined as internal at any time during the IVR.
6. Select a Group. Each group lets you define a sequence of steps. Define the steps for “Main” first.
- Define steps for other groups as needed. The groups are called as follows:
- Main: Called when the IVR is activated.
 - 0 1 2 3 4 5 6 7 8 9 * #: Called during the IVR when a user presses a key on their phone.
 - i: Called when a user makes an invalid choice.
 - t: Called when the ResponseTimeout expires. See “ResponseTimeout” in Section 7.5.3.

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7. Under Step, select "Ringing" as the first step in the IVR. This provides feedback to the caller that the line has connected.
8. For the next step, select "Wait" and set a value of "2." This sets ringing to continue for 2 seconds.
9. For the next step, select "Answer" to pickup the call.
10. For the next step, select "Background" and then select IVR-welcome.gsm. This plays the default welcome message.
11. For the next step, select "WaitExten" and set a value of "1." This inserts a short pause, but allows calls to enter input.
12. For the next step, select "Background" and then select "VR-if-u-know-ext.gsm." This plays a message asking the user to enter the extension of the person they want to reach. The user can press the extension number (followed by the # sign) while the message is playing. You can record your own message and assign it here. See "Recording voice prompts" in Section 7.10.
13. For the next step, select "WaitExten" and set a value of "3." This inserts a pause allowing the caller to enter input.
14. At this point, the IVR looks like this:

Step	Parameter	Action
Ringing		Delete Up Down
Wait	2	Delete Up Down
Answer		Delete Up Down
Background	IVR-welcome.gsm	Delete Up Down
WaitExten	1	Delete Up Down
Background	IVR-if-u-know-ext.gsm	Delete Up Down
WaitExten	3	Delete Up Down

Figure 7-53. IVR screen.

15. Create additional steps as required.
 - Refer to "IVR commands" in Section 7.5.3 for a description of all available commands and their parameters.
 - Use the "Delete," "Up," and "Down" buttons to adjust the order of the steps.
 - Blank steps are ignored.
16. Click "Save" when you are done. The new IVR will be added to the list on the IVR Menus page.
17. Click "Apply Configuration" to activate the IVR.

7.5.3 IVR Commands

Answer

Answers an incoming call.

Parameter: None.

Authenticate

Prompts the caller to supply a numeric password to continue.

Parameter: Specify a password that is between 5 and 8 digits in length.

Background

Plays a sound file while waiting for input from the caller. Playback ends as soon as the caller presses a key.

Parameter: Select a predefined sound file. To add sounds file to the Hybrid PBX and VoIP Gateway, or record a custom message, see "Recording voice prompts" in Section 7.10.

Busy

Plays a busy tone until the caller hangs up. Subsequent steps are not executed.

Parameter: None.

CheckVoicemail

Prompts the caller to enter a voicemail extension number to retrieve messages. After the caller enters the extension, the caller is then prompted for the voicemail password.

Parameter: None.

Congestion

Plays a fast busy tone until the caller hangs up. Subsequent steps are not executed.

Parameter: None.

Dial

Dials the specified phone number using the selected line.

Parameter 1: Select the outgoing line group.

Parameter 2: Specify the phone number to dial.

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DigitTimeout

Sets the maximum amount of time between key presses when a caller is entering data.

Parameter: Maximum time between digits in seconds.

DISA

Direct Inward System Access (DISA). The caller is prompted for a password, after which the call is terminated. The caller's password and CallerID is checked against the DISA list. If valid, Hybrid PBX and VoIP Gateway calls the caller and provides them with a dial tone. The caller can then dial out through the Hybrid PBX and VoIP Gateway PBX using the specified outgoing line group and call rules. See "Configuring DISA users" in Section 7.8.

Parameter 1: Outgoing Call Rules

Parameter 2: Outgoing Line Group

DISA-NoPassword

Same functionality as the DISA command, except that instead of prompting for a password, the caller is validated against the DISA list using their CallerID.

Parameter 1: Outgoing Call Rules

Parameter 2: Outgoing Line Group

GotoDirectory

The caller is prompted to enter the first three characters of the user name they want to reach using the telephone keypad. The Hybrid PBX and VoIP Gateway then plays a list of matching users from which the user can select.

Parameter: Select "LastName" or "FirstName" to define how names are searched.

Example: If a caller enters "7,6,4," a search is done for names beginning with "Smi."

GotoExtension

Sends the caller to the specified extension, which can be a SIP extension, an analog extension, an IVR extension, a virtual voicemail box, a conference room, or a ring group.

Parameter: Select an extension.

GotoMenu

Sends the caller to the top of the specified IVR menu.

Parameter: Select an IVR menu name.

GotoTimebasedRule

Transfers control to the specified IVR or IVR label if the selected time frame is currently in effect.

Parameter 1: Name of time frame to be validated.

Parameter 2: Select the IVR to which control will be transferred.

Hangup

Terminate call.

Parameter: None.

Label

Tags the previous step with the specified label name. Allows entry into an IVR at a specific step.

Generally used to skip a number of preliminary steps in an IVR.

Parameter: Specify a name to identify the label.

Language

Sets the language for all system level prompts played to the caller by this IVR.

Parameter: Select a language.

LeaveVoicemail

Sends a caller to the specified voicemail box.

Parameter: Select a voicemail box.

Playback

Plays a sound file to completion, ignoring any caller input. When playback of the file is complete, control moves to the next step in the IVR.

Parameter: Select the sound file to play.

PlayInvalid

Plays a system sound file to advise caller that they have made an invalid entry.

Parameter: None.

ResponseTimeout

Defines the maximum amount of time during which the caller must make a selection or enter an extension number. If this time is exceeded, then the steps associated with the group are executed. See Section 7.5.2.

Parameter: Specify the maximum response time in seconds.

Ring

Plays a ringing sound for the specified number of seconds. Generally used in combination with Wait.

Parameter: Specify ringing time in seconds.

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SetCallerID

The caller's callerID is modified with the specified prefix or suffix. If Parameter 1 starts with a number, then the callerID number is modified. Otherwise, the CallerID name is modified.

Parameter 1: Characters to add to the callerID.

Parameter 2: Select either "Prefix" or "Suffix."

SetMusicOnHold

Specifies the music that is played when a caller is on hold. This can be a sound file or the Audio In port (Audio In not available on the G-122). See "Setting up music on hold" in Section 6.8.

Parameter: Select a sound file, or DefaultAudioIn to use the Audio In port.

Wait

Pauses execution for the specified number of seconds before moving to the next step. During the wait, caller input is ignored. Generally used after Ringing.

Parameter: Time to wait in seconds.

WaitExten

Pauses execution for the specified number of seconds before moving to the next step. Caller input is processed while waiting.

Parameter: Time to wait in seconds.

7.6 Setting Up Time Frames

Time frames work in conjunction with the GotoTimeBasedRule IVR command (see Section 7.5.3), giving you a way to customize IVR menu behavior based on the current system date and time.

For example, you could create a time frame called AfterHours that defines the times during which your office is closed. By checking this time frame in an IVR menu, you could play a specific prompt or route calls to a specific voicemail box when your office is closed.

7.6.1 Permanent vs. OnDemand Time Frames

Two types of time frames can be created:

- A permanent time frame is active during the range of dates and times that you configure.
- An OnDemand time frame is only active after you dial a specific extension number and enter a password. OnDemand time frames are useful when you do not have a fixed schedule for activating an IVR. For example, if you have a flexible lunch hour, or opening/closing hours, you can activate time frames as needed without being tied to a fixed schedule.

7.6.2 Creating a Time Frame

1. Select "PBX > Incoming Calls > Time Frames." The Time Frames page opens.



Figure 7-54. Time frames page.

This page shows all time frames that are defined in the system. The following information is displayed for each time frame.

- Name: Name of the time frame.
 - Action: Click the appropriate button to “Edit” or “Delete” a time frame. The “Edit” button displays the Time Frames page, which allows you to edit an existing time frame.
2. Click “Add.” The Time Frames page opens.

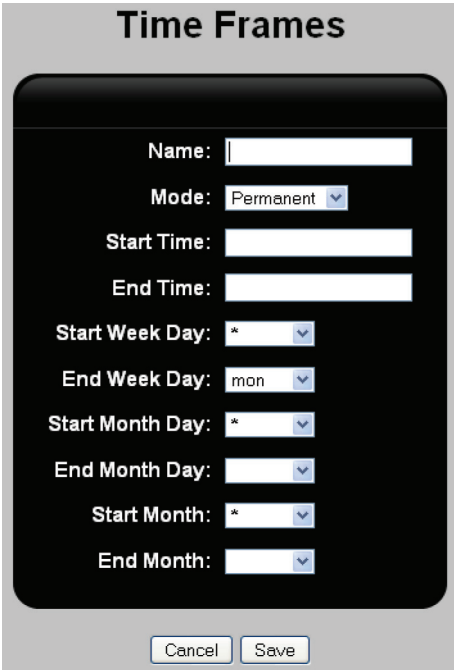
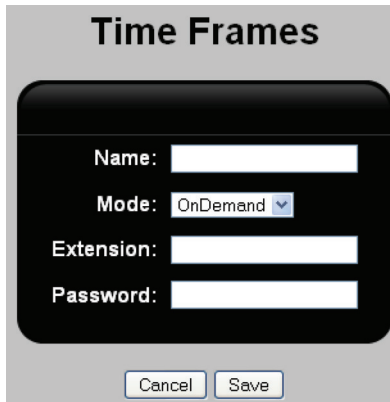


Figure 7-55. Time frames page.

3. For Name, specify a name to identify the time frame.
4. For Mode, select one of the following:
- Permanent: Select this option to define a time frame that follows a fixed schedule.

- OnDemand: Select this option to define a time frame to be activated on demand by dialing an extension and specifying a password.

5. If you selected "OnDemand" for Mode, configure the following settings:

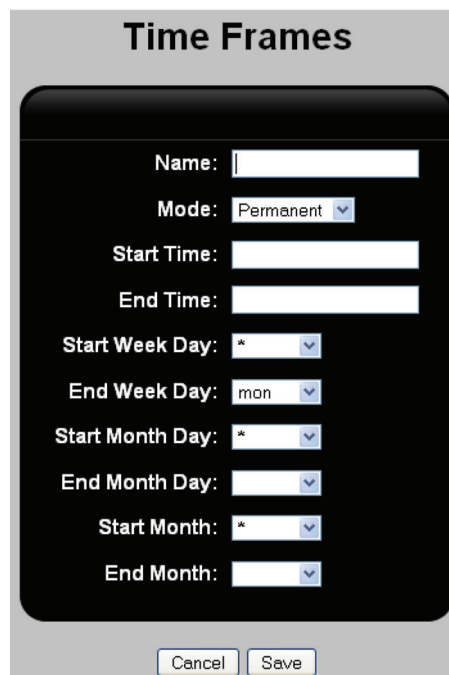


The image shows a configuration dialog titled "Time Frames". It has a dark background with white text and input fields. The "Mode" dropdown is set to "OnDemand". The fields are: Name (text input), Mode (dropdown menu showing "OnDemand"), Extension (text input), Password (text input), and buttons for "Cancel" and "Save" at the bottom.

Figure 7-56. Time frames OnDemand mode selected.

- Extension: Specify the extension number that will activate the time frame.
- Password: Specify the password (numbers only) that will activate the time frame.

6. If you selected "Permanent" for Mode, configure the following settings:



The image shows a configuration dialog titled "Time Frames". It has a dark background with white text and input fields. The "Mode" dropdown is set to "Permanent". The fields are: Name (text input), Mode (dropdown menu showing "Permanent"), Start Time (text input), End Time (text input), Start Week Day (dropdown menu showing "*"), End Week Day (dropdown menu showing "mon"), Start Month Day (dropdown menu showing "*"), End Month Day (text input), Start Month (dropdown menu showing "*"), End Month (text input), and buttons for "Cancel" and "Save" at the bottom.

Figure 7-57. Time frames permanent mode selected.

- Start Time: Time in the format: hh:mm:ss. Specify hh as 0–24. Leave blank to specify midnight.
- End Time: Time in the format: hh:mm:ss. Specify hh as 0–24. Leave blank to specify midnight.
- Start Week Day: Select "*" to select every day of the week.

- End Week Day: Select the ending week day.
- Start Month Day: Select “*” to select every day of the month.
- End Month Day: Select the ending day.
- Start Month: Select “*” to select every month of the year.
- End Month: Select the ending month.

7. Click “Save” when you are done. The new time frame will be added to the list on the Time Frames page.

8. Click “Apply Configuration” to activate the time frame.

7.7 Creating Park and Page Extensions

The park and page feature is designed for businesses that want to personally answer all incoming calls, but may not always be immediately available when the phone rings. When configured, park and page automatically places an incoming call into a parking room and plays the defined welcome message. It then plays a paging message (on the Audio out port) to alert someone in the office to pick up the phone. If the call is not answered within a specific time period, the call can then be routed to a voicemail box, IVR, or ring group.

To create a park and page extension

1. Create two custom messages: one to welcome callers, and another to play over the paging system to alert staff when a call arrives. See “Recording voice prompts” in Section 7.10.
2. Select “PBX > Incoming Calls > Park and Page.” The Park and Page page opens.



Figure 7-58. Park and page screen.

3. Click “Add.” The Park and Page page opens.




The image shows a configuration window titled "Park and Page". It contains several fields: "Extension:" with an empty text box, "Name:" with an empty text box, "Welcome Message:" with a dropdown menu showing "En-AfterHours.gsm", "Park Message:" with a dropdown menu showing "En-AfterHours.gsm", "Park Time:" with an empty text box, and "Timeout Extension" with a dropdown menu showing "6000". At the bottom are "Cancel" and "Save" buttons.

Figure 7-59. Park and page screen.

Configure settings as follows:

- Extension: Specify an unused extension number.
- Name: Specify a name to identify this park and page extension.
- Welcome Message: Select the sound file that will play after the call is parked.
- Park message: Select the sound file that will play on the Audio out port to alert users to the incoming call.
- Park time: Specify how long the caller will be kept on hold before being forwarded to the Timeout Extension.
- Timeout Extension: Select the destination to which the call will be forwarded when Park Time expires. The list lets you choose from all users, voicemail boxes, IVRs, and ring groups. For example:



The image shows the "Park and Page" configuration window with the following values: "Extension:" is "9000", "Name:" is "Park-Page", "Welcome Message:" is "Park-Page_Welcome.gsm", "Park Message:" is "Park-Page_Welcome.gsm", "Park Time:" is "20", and "Timeout Extension" is "Box1". Red arrows point from the text "Custom prompts." to the "Welcome Message" and "Park Message" dropdowns, and from the text "If not answered, go to voicemail box." to the "Timeout Extension" dropdown. "Cancel" and "Save" buttons are at the bottom.

Figure 7-60. Adding custom prompts to park and page screen.

4. Click "Save" when you are done. The new entry will be added to the list on the Park and Page page.

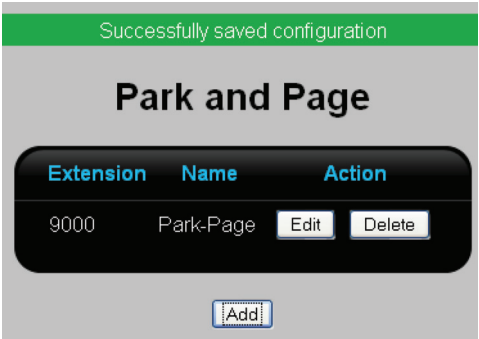


Figure 7-61. Park and page screen.

5. Click “Apply Configuration” to activate the new park and page entry.

7.8 Configuring DISA Users

The DISA (Direct Inward System Access) feature enables external callers to dial into the Hybrid PBX and VoIP Gateway and then dial out on any line. DISA protects inbound access using either a password or by validating the caller’s callerID.

7.8.1 Using DISA

- First add one or more DISA users as shown below.
- Next, create an IVR with one of the following commands: DISA or DISA-NoPassword. See “Creating Interactive Voice Response (IVR) menus” in Section 7.5.2.

7.8.2 Adding a DISA User

1. Select “PBX > Incoming Call Features > DISA.” The DISA page opens. Initially, it is empty.



Figure 7-62. Empty DISA page.

2. Select “Add.” The DISA page opens.



Figure 7-63. DISA page.

3. Configure the parameters as follows.

- Name: Specify the name of the user.
- Phone number: Specify the phone number that will be used to validate the user’s identity.
- Password: Specify the password the user must enter when prompted (when you use the DISA IVR command).
- Account code: Specify an accounting code to identify the user of this account. In most cases this would be set to the extension number of the user. This code appears in the accountcode column in the call detail record. See “Tracking call information” in Section 7.13.
- CallerID Name: Specify the user’s CallerID name that will be used when the user completes a call through the PBX after being validated. This information is also displayed when the Hybrid PBX and VoIP Gateway PBX calls back to validate the user.
- CallerID Number: Specify the CallerID number that will be used when the user completes a call through the PBX after being validated.

4. Click “Save.”

5. Click “Apply Configuration.”

7.9 Configuring Ring Groups

A ring group defines an extension number that when dialed, rings one or more users simultaneously or in succession. A common use for a ring group is to ring all users in a team (like a sales group), for every incoming call, or to replace an automated attendant in a small office.

Creating a ring group

1. Select “PBX > Incoming Calls > Ring Groups.” The Ring Groups page opens.



Figure 7-64. Empty ring groups page.

This page shows all ring groups that are defined in the system. Initially it is empty. The following information is displayed for each ring group.

- Extension: Extension number assigned to the ring group.
- Name: Name assigned to the ring group for identification purposes.
- Action: Click the appropriate button to Edit or Delete a ring group. The Edit button displays the Ring Groups page which allows you to edit an existing ring group.

2. Click “Add.” The Ring Groups page opens.

A screenshot of the 'Ring Groups' page showing the configuration form for a new ring group. The form includes the following fields: 'Extension' (text input), 'Name' (text input), 'Ring Time' (dropdown menu with '5' selected), 'Play Back' (dropdown menu with 'Ring' selected), 'Strategy' (dropdown menu with 'Ring All' selected), 'Ring All Extensions' (checkbox), 'Ring Paging Speaker' (checkbox), 'User' (dropdown menu), 'Phone Number / Outgoing Line Groups' (text input), 'Action' (text input), and 'If not Answered' (dropdown menu with 'Hangup' selected). At the bottom of the form, there are 'Cancel' and 'Save' buttons.

Figure 7-65. Ring groups page.

Configure the parameters as follows.

- Extension: Specify a unique extension number for the ring group. Extensions must be between three and six digits in length.
- Name: Specify a name to identify the ring group.

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- Ring Time: Specify how many seconds to ring an extension before moving on to the next extension if the call is not answered. If Ring All Extensions is enabled, the caller is sent to the destination set for "If not Answered."
 - Play back: Specify what the caller hears while the call is waiting to be answered. Normally this is set to Ring, which plays a ring tone giving the caller the impression that the call is ringing. Other options include the music on hold files located in PBX > Sound Manager > Sound Manager > Music on Hold.
 - Strategy: Select a ring strategy.
 - Ring All: Ring all user extensions in the ring group at the same time.
 - Ring in Order: Ring each extension in the ring group sequentially.
3. Select Ring All Extensions to make all extensions defined on the Hybrid PBX and VoIP Gateway part of this ring group.
 4. Select Ring Paging Speaker to have a ring tone sent on the Audio out port.
 5. If Ring All Extensions was not selected, define the members of the ring group.
 - User: Select a user to add to the ring group.
 - Phone Number/Outgoing Line Groups: (Not supported on analog lines.) If you select the External option under User, these fields let you define an external phone number to reach the user. For example, this could be the user's home phone or cell number. You also need to select which outgoing line group the Hybrid PBX and VoIP Gateway will use to make the call.
 - Action: Click "Delete" to remove the user from the ring group.
 6. Click "Save" when you are done. The ring group will be added to the list on the Ring Groups page.
 7. Click "Apply Configuration" to activate the new ring group.

7.10 Recording Voice Prompts

Many of the call management features make use of voice prompts. A number of prompts are supplied with the Hybrid PBX and VoIP Gateway Gateway, but in many cases you will want to create your own.

7.10.1 Direct Recording with a Phone

The fastest way to record voice prompts is to use a phone connected to the Hybrid PBX and VoIP Gateway.

1. Dial "*95" from any phone and wait for the beep.
2. Record your message. Press "#" when done. You will then be given the option to press "1" to listen to your message, or press "2" to re-record your message.
3. Hangup the phone.
4. Select "PBX > Sound Manager > Sound Manager > IVR." You will see a file starting with the extension number of the phone you used to record the message. For example, the following screen shows two recordings made with the phone at extension 6002.

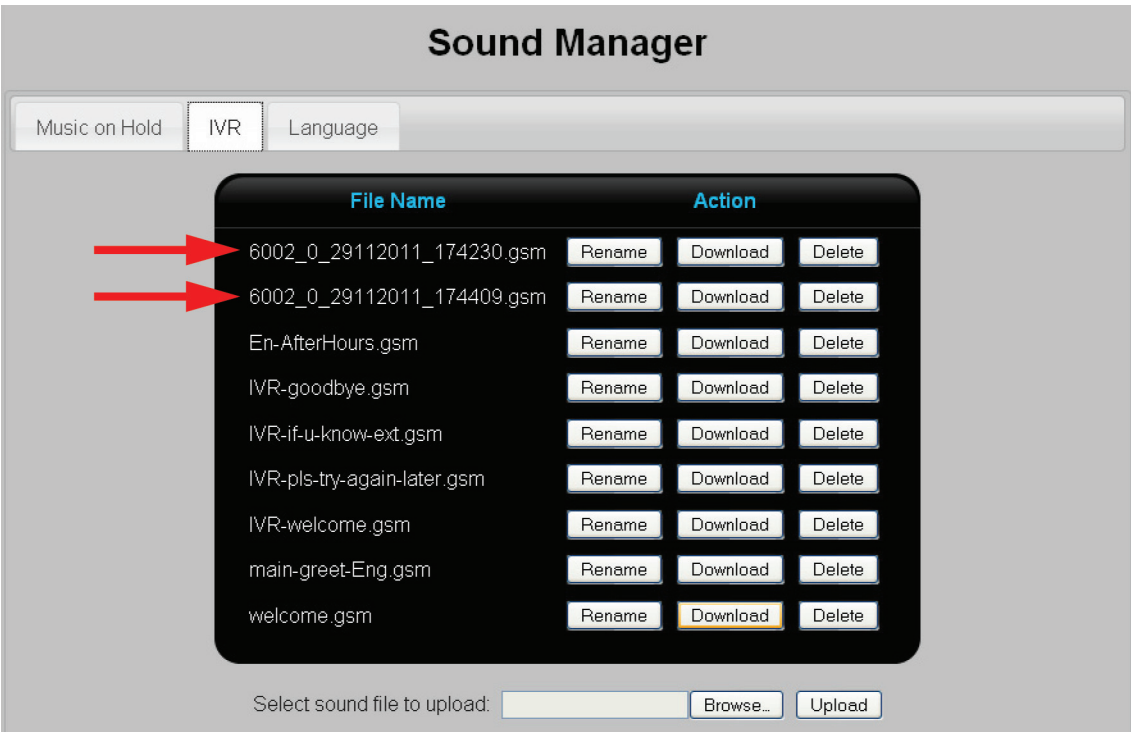


Figure 7-66. Sound manager screen.

5. Click "Rename" for each file and give it a new, meaningful name.

- Do not use spaces in the file name.
- Do not change the file extension.

For example, if these files were going to be used for park and page we might rename them to Park-Page_Welcome.gsm and Park-Page_Paging.gsm.

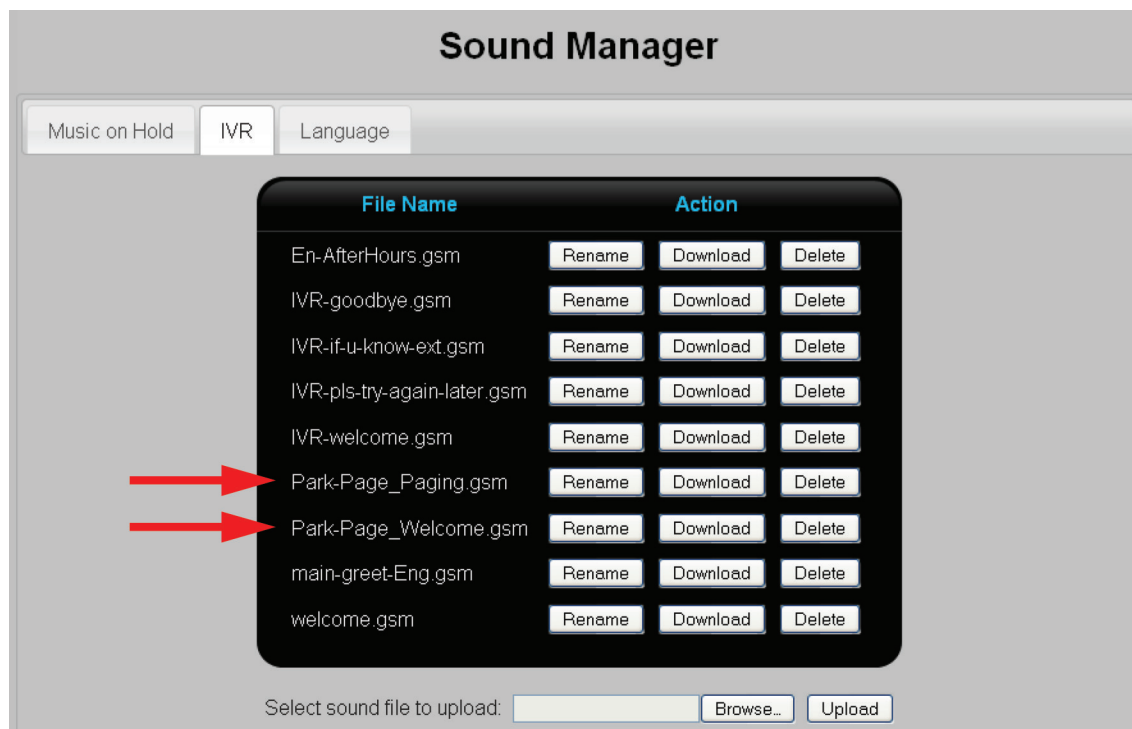


Figure 7-67. Sound manager screen.

7.10.2 Uploading Sound Files

Another way to add voice recordings is to upload them to the Hybrid PBX and VoIP Gateway. This gives you the flexibility to fine tune the recordings using third-party software.

NOTE: Sound files must be converted to gsm format before you upload them.

1. Record music on your computer to a sound file in gsm format. To convert a stereo "wav" file to "gsm," you can use Sox (freely available on the Internet) with the following command:

```
sox music.wav -r 8000 -c1 music.gsm resample -q1
```

For better results, try recording the wave file in 16-bit mono at a sampling rate of 8000 Hz and then run:

```
sox music.wav music.gsm
```

(Another useful program is the Switch Sound File Converter, which is also available as a free download.)

2. Click "Browse" at the bottom of the IVR page and select the sound file on your computer.
3. Click "Upload" to send it to the Hybrid PBX and VoIP Gateway.

7.11 Creating a Virtual Conference Room

A virtual conference room lets multiple callers talk to each other at the same time. You can access the room by directly dialing an extension, or through an IVR.

Conference rooms are selected by extension number and protected with both administrator and participant passwords.

NOTE: By default, the maximum number of conference rooms is limited to two. To change this, select "PBX > PBX Settings > General."

7.11.1 Viewing All Conference Rooms

Select “PBX > Conference Bridge.” The Conference Bridge page opens.

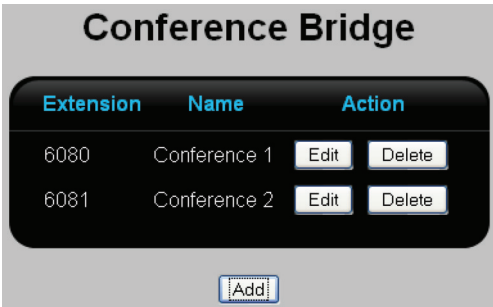


Figure 7-68. Conference bridge page.

This page lists all defined conference rooms. The following information is presented for each conference room.

- Extension: Extension number to reach the conference room.
- Name: Name that identifies the conference rooms.
- Action: Click the appropriate button to “Edit” or “Delete” a room. The “Edit” button displays the Conference Bridge edit page.
- Add button: To add a new conference room, click Add, then see Section 7.11.2.

7.11.2 Creating a Conference Room

1. Click “Add” on the Conference Bridge list page. This displays the Conference Bridge edit page.

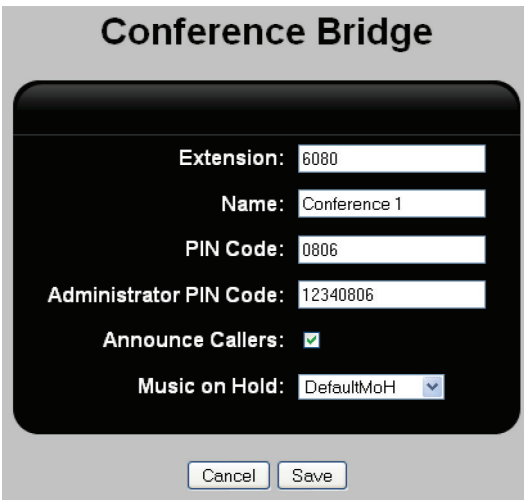


Figure 7-69. Conference bridge edit page.

2. Configure the following settings:

- Extension: Specify the extension number that users will dial to reach the conference room.
- Name: Specify a descriptive name for the conference room.
- PIN Code: This code is given to participants in the conference. Conferees must enter the PIN code to access the conference.
- Administrator PIN Code: Used by the conference administrator to enter the room. The conference administrator has access to the following room management functions by dialing “*:”

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- 1: Mute/Unmute.
 - 2: Lock/Unlock the conference room.
 - 3: Eject last user who joined the conference room.
 - 4: Decrease conference room volume each time "4" is pressed. End with "8."
 - 6: Increase conference room volume each time "6" is pressed. End with "8."
 - 7: Decrease personal volume each time "7" is pressed. End with "8."
 - 9: Increase personal volume each time "9" is pressed. End with "8."
- Announce Callers: When enabled, after participants login with a valid password they are prompted to say their name and press the # key. This recording is then played to the participants already in the conference when the new member is added.
 - Music on Hold: Select the music on hold to play while a participant is on hold waiting for another participant to join. See "Setting up music on hold" in Section 6.8.
3. Click "Save" when you are done.
 4. Click "Apply Configuration."

7.12 Mapping External Phone Numbers to Extensions

The phone book feature lets you map external phone numbers to extensions, making them easy to dial for all internal users. This is a great way to simplify the dialing of long phone numbers that are frequently used by many people in an organization. It also is a good way to force certain numbers to be dialed with a specific outgoing call rule.

7.12.1 Viewing All External Phone Number Mappings

Select "PBX > Phone Book." The Phone Book list page opens.

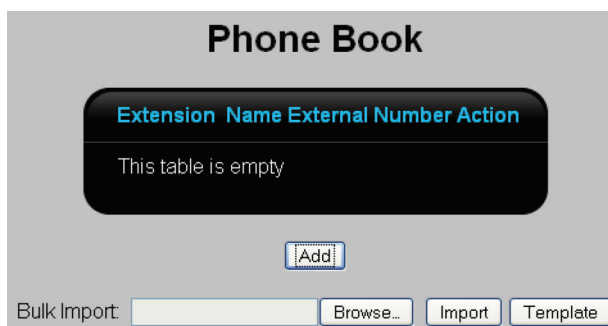


Figure 7-70. Phone book list page.

This page lists all defined external phone mappings. The following information is presented for each mapping.

- Extension: Extension that internal user's dial to dial an external number.
- Name: Unique name that identifies the mapping.
- External number: External number that will be dialed.
- Action: Click the appropriate button to "Edit" or "Delete" a mapping. The "Edit" button displays the Phone Book edit page. See "Mapping a phone number" in Section 7.12.2.
- Add button: To add a new mapping, click "Add," then see "Mapping a phone number" in Section 7.12.2.

7.12.2 Mapping a Phone Number

1. Click “Add” on the Phone Book list page. This displays the Phone Book edit page.

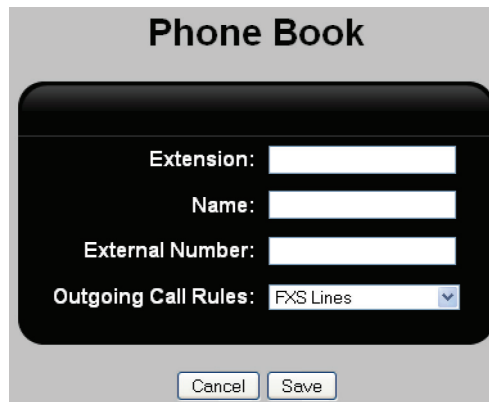
The image shows a web form titled "Phone Book" with a dark background. It contains four input fields: "Extension:" (empty), "Name:" (empty), "External Number:" (empty), and "Outgoing Call Rules:" (a dropdown menu showing "FXS Lines"). At the bottom are "Cancel" and "Save" buttons.

Figure 7-71. Phone book page.

2. Configure the following settings:

- Extension: Specify the extension that internal users dial to dial the external number.
- Name: Specify a unique name to identify the mapping.
- External number: Specify the external number to dial. Make sure to add any dialing prefixes as specified in the outgoing call rule you select.
- Outgoing Call Rules: Select the rule that will apply to all calls using this mapping.

For example:

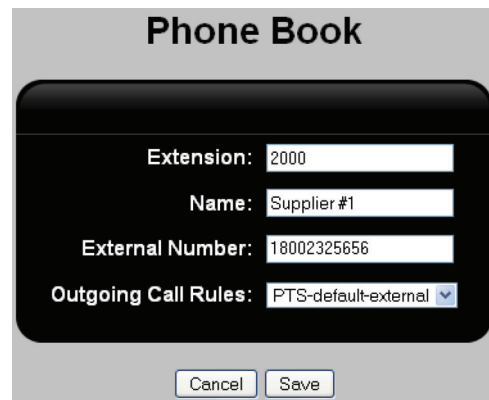
The image shows the same "Phone Book" form as Figure 7-71, but with example data entered. The "Extension:" field contains "2000", the "Name:" field contains "Supplier #1", the "External Number:" field contains "18002325656", and the "Outgoing Call Rules:" dropdown menu shows "PTS-default-external". The "Cancel" and "Save" buttons are at the bottom.

Figure 7-72. Phone book screen example.

3. Click “Save.” The Phone Book list page opens showing the new mapping. For example:

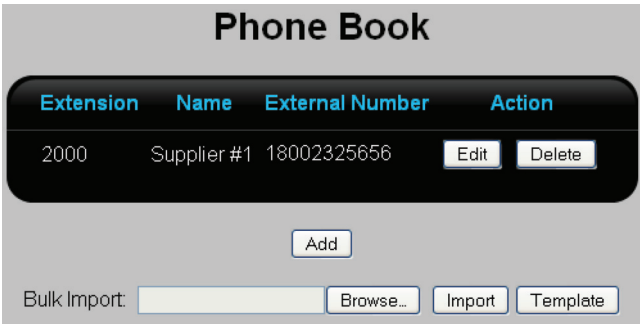


Figure 7-73. Phone book list page, new mapping.

4. Click "Apply Configuration" to activate the new mapping.

7.13 Tracking Call Information

The Hybrid PBX and VoIP Gateway logs information for all internal and external calls made through the PBX. You can view this information with command center, or download it for processing with other applications. The downloaded version contains additional information which is not displayed in command center.

7.13.1 Viewing Call Information

Select "System > CDR." The CDR page opens.

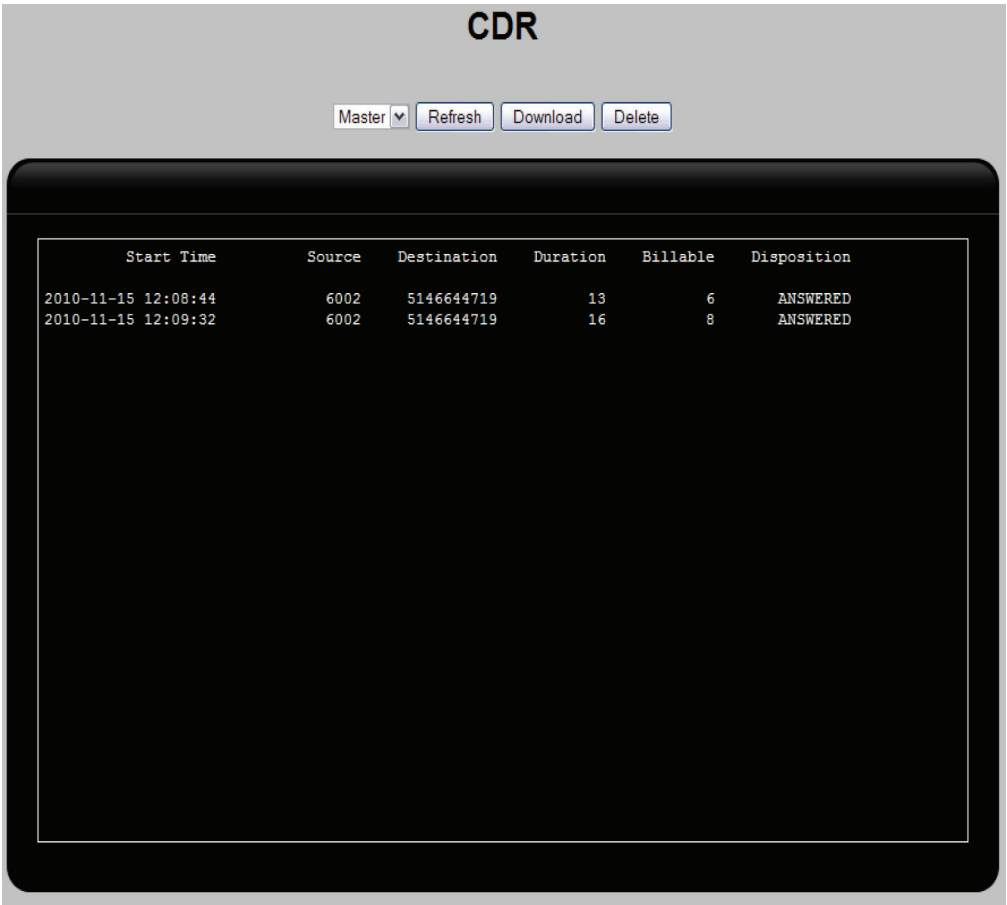


Figure 7-74. CDR page.

This page displays information from the Call Detail Record (CDR) system. Buttons allow refreshing the display and downloading the CDR data as a “tar.gz” file.

- Start Time: Start time of the call.
- Source: Source telephone or extension number.
- Destination: Number of the called party.
- Duration: Total duration of the call including ring time in seconds.
- Billable: Duration of the call from answer to hang up, excluding ring time, in seconds.
- Disposition: Call completion information: ANSWERED, NO ANSWER, BUSY, FAILED.
- Accounting: Only visible when accounting is enabled (PBX > PBX settings > Call Handling > Call Accounting). When enabled, the accounting code used to make a specific call is shown.

7.13.2 Downloading Call Data Information

Click the “Download” button to save the CDR information to your computer in a “tar.gz” file.

This file contains the following columns (in order):

- accountcode: Billing account number (string, 20 characters).
- src: CallerID number (string, 80 characters).
- dst: Number of the called party (string, 80 characters).
- dcontext: Destination context (string, 80 characters).
- clid: CallerID with text (80 characters).
- channel: Channel used (80 characters).
- dstchannel: Destination channel if appropriate (80 characters).
- lastapp: Last application if appropriate (80 characters).
- lastdata: Last application data (arguments) (80 characters).
- start: Start time of the call.
- answer: Date/Time the call was answered (date/time).
- end: End of call (date/time).
- duration: Total duration of the call including ring time. in seconds (integer).
- billsec: Duration of the call from answer to hang up, excluding ring time, in seconds (integer).
- disposition: Call completion information: ANSWERED, NO ANSWER, BUSY, FAILED.
- amaflags: Indicates what flags to use: DOCUMENTATION, BILL, IGNORE. Default DOCUMENTATION
- UniqueID: Unique Channel Identifier.

7.14 Configuring Global Call Handling Settings

Select “PBX > PBX Settings > Call Handling” to configure global settings for various call management features.

The image shows a web-based PBX settings interface. At the top, there's a title "PBX Settings" and a series of tabs: "General", "Call Handling", "SIP", "Voicemail", "Recordings", "Outlook", and "Trunks / Lines". The "Call Handling" tab is currently selected. Below the tabs, there's a large black rounded rectangle containing a list of settings, each with a label and a text input field. The settings are: "Blind Transfer:" with value "#1", "Warm Transfer:" with value "#2", "Pick-up:" with value "#7", "Voicemail Extension:" with value "6050", "Remote Voicemail Extension:" with value "6051", "IP Address Extension:" (empty), "IVR Recording Extension:" with value "*95", "Spy Extension:" (empty), "Spy Password:" (empty), "Audio-In Extension:" with value "*97", "Paging Extension:" with value "*96", "Desktop (Phone) Paging Extension:" with value "*94", "Parking Extension:" with value "700", "Parking Rooms (From):" with value "701", "Parking Rooms (To):" with value "720", "Parking Time:" with value "45", "Cell Phone Prefix:" with value "7", "BLF Pickup Prefix:" with value "*7", "Extension Paging Prefix:" (empty), "Voicemail Prefix:" (empty), and "Call Accounting:" with a checked checkbox. At the bottom of the black rectangle is a "Save" button.

Setting	Value
Blind Transfer:	#1
Warm Transfer:	#2
Pick-up:	#7
Voicemail Extension:	6050
Remote Voicemail Extension:	6051
IP Address Extension:	
IVR Recording Extension:	*95
Spy Extension:	
Spy Password:	
Audio-In Extension:	*97
Paging Extension:	*96
Desktop (Phone) Paging Extension:	*94
Parking Extension:	700
Parking Rooms (From):	701
Parking Rooms (To):	720
Parking Time:	45
Cell Phone Prefix:	7
BLF Pickup Prefix:	*7
Extension Paging Prefix:	
Voicemail Prefix:	
Call Accounting:	<input checked="" type="checkbox"/>

Figure 7-75. PBX settings screen.

Configure settings as follows:

- **Blind Transfer:** Specify the prefix to dial when transferring a call to another extension without warning the recipient (also called a cold transfer). The default prefix is #1. To transfer a call, the user dials #1 followed by the extension and then hangs up.
- **Warm Transfer:** Specify the prefix to dial when transferring a call to another extension after warning the recipient. The default prefix is #2. To transfer a call, the user dials #2 followed by the extension, talks to the recipient, and then hangs up.
- **Pick-up:** Specify the key to press to pickup a call. The default is: #7
- **Voicemail Extension:** Extension to specify when users call in to check their personal voicemail messages. Default: 6050. See "Voicemail configuration" in Section 8.1 and "Managing voicemail" in Section 8.3.

- Remote Voicemail Extension: Extension to be used when users call in to check their voicemail messages from a different phone. This can also be used to check voicemail when calling in on an external line if there is an IVR handling the calls. Dialing this extension during the IVR gives the user access to the voicemail system. See “Voicemail configuration” in Section 8.1 and “Managing voicemail” in Section 8.3.
- IP Address Extension: Extension to be dialed when an administrator needs to find the network IP address of the Hybrid PBX and VoIP Gateway. This feature works only from the analog phones connected to the FXS ports.

To retrieve the IP Address of a Hybrid PBX and VoIP Gateway Gateway:

1. Dial the IP Address Extension. If an IP Address Extension has not been set, then dial ##**
 2. Enter the password: 1234567890# (cannot be changed).
 3. The IP Address will be presented.
- IVR Recording Extension: Local digits to dial to access the built-in voice recorder for IVR menus. The default value is “*95.” See “Recording voice prompts” in Section 7.10.
 - Spy Extension: Extension to dial to initiate call monitoring on a selected user. “Monitoring calls” in Section 6.7. Default: None
 - Spy Password: Password to specify to initiate call monitoring. Default: None
 - Audio-In Extension: Extension to dial to allow a user to listen to the audio source connected to the Audio In port. Default: *97.
 - Paging Extension: Extension to dial to allow a user to speak over the Audio Out port. This port is generally connected to a paging system. Default: *96.
 - Desktop (Phone) Paging Extension: (Available on phones with multicast support, such as Snom.) Extension to dial to activate the paging system on phones that have local speakers. Default: *94
 - Parking Extension: The park extension assigns a parking room to a call that is taken from one extension, is then put on “hold” only to be taken from another extension (usually by the called party). The parking room that is assigned is an extension within the range specified by “Parking Rooms.” Default: 700

When a call is parked, the parking room is announced to the user putting the call into park. This way the user will know where to retrieve the call.

This feature also works well with BLF. A user can configure BLF to monitor parking rooms 701-720, so they'll know when a caller is waiting and in what room.

- Parking Rooms: Range of extensions that can be assigned to parking rooms. Default: 701–720
- Parking Time: Number of seconds of park time allowed. After this time the calls are routed back to the original extension. Default: 45 (seconds)

To park a call then resume the call from another extension: (In this example, the number for a blind transfer is #1, and the code for the parking extension is 700. The range 701–720 is available for parking rooms.)

1. While on a call, dial #1 followed by 700. The automated attendant will assign a parking room (in this example) of 705.
2. Hang up.
3. Pick up any other extension.
4. Dial “705” (the assigned parking room).
5. Resume the call.

If the call was not answered within the number of seconds specified in Parking Time above, the call will be routed back to the original extension.

- Cell Phone Prefix: Prefix to dial to reach the cell phone number configured for a user. Used in combination with the user’s extension number. For example, to dial the cell phone of extension 6565, you would dial “76565.” Default: 7

Chapter 7: Call Management

- **BLF Pickup Prefix:** Specify the prefix that can be used to pickup a call on an extension that is flashing the Busy Lamp Field (BLF). The prefix can also be used to speed dial the extension as well. For example, if using BLF for extension 6005, not only would you be able to pick up the calls to that extension when monitoring the assigned button, but the button also doubles as a speed dial to 6005.

This is very useful when working with the park and page feature, as the user would be able to tell if there is anyone parked and waiting, and would be able to (speed) dial the park room simply by pressing the button assigned.

The BLF is provisioned by selecting "PBX > Users > Phone Provisioning." Default: *7.

For example, the user on extension 6012 has a BLF which shows extension 6011 is flashing and has not been answered. The user picks up their phone and dials *76011 to answer the call.

If *76011 is put into the pickup value field under BLF in the provisioning field of extension 6012, that extension can be picked up by pressing the button associated with the flashing lamp.

- **Extension Paging Prefix:** Prefix which allows paging a particular extension's speaker. Default: *9.

For example, to page the user at extension 6005, dial "*96005" and speak.

- **Voicemail Prefix:** Allows a message to be left in the extension's voicemail box without ringing the extension. If the value of *6 is entered then a call to *66011 would leave a message in the voicemail box of extension 6011 without ringing the extension.

*NOTE: *6 is only an example, there is no default value.*

- **Call Accounting:** Enables call accounting. See "Configuring call accounting" in Section 7.4

7.15 Call Recordings

The call recordings feature enables users to make a recording of any phone conversation.

7.15.1 Setting Up Call Recording

Configure call recording support and storage location.

1. Select "System > Maintenance > Logs."



The screenshot shows a web interface titled "Maintenance". At the top, there is a navigation bar with tabs: "Date and Time", "Firmware Upgrade", "Reset to Defaults", "Backups", "Storage", "Logs", and "Dial-in". The "Logs" tab is selected. Below the navigation bar, there is a central configuration panel with a black background and white text. The panel contains the following settings:

- CDR:** Disabled (dropdown menu)
- Recordings:** Disabled (dropdown menu)
- Voicemail:** Disabled (dropdown menu)
- Email Log:** RAM (dropdown menu)
- GUI Log:** RAM (dropdown menu)
- Web Server Log:** RAM (dropdown menu)

At the bottom of the configuration panel, there is a "Save" button.

Figure 7-76. Maintenance screen.

2. For Voicemail, select a storage option. This also enables voicemail support.

NOTE: You must restart the Hybrid PBX and VoIP Gateway for this setting to become active.

NOTE: If USB or Compact Flash storage is specified, but not installed, the voicemail will be stored in system memory, and it will be lost when the system restarts.

3. Click "Save."

Configuring global call recording settings

1. Select "PBX > PBX Settings > Recording."



Figure 7-77. PBX settings, recording.

2. For Record Calls, select a recording option:

- Never: Call recordings are disabled.
- OnDemand: Users can start and stop call recording by dialing the recording prefixes.
- Always: Every call on the Hybrid PBX and VoIP Gateway is recorded.

3. For Start OnDemand Recording: Specify the key combination that a user enters to start recording a call.

4. For Stop OnDemand Recording: Select the key combination that a user enters to stop recording a call.

5. Click "Save."

6. Click "Apply Configuration."

Configuring user account settings

For each user that will make recordings, you need to enable the Show Recordings option. See Section 6.2.1 for SIP users and Section 6.2.3 for analog users.

7.15.2 Making a Call Recording

If OnDemand call recording is enabled, then:

- To start a recording, the user presses the start key combination. By default, this is #8.
- To end a recording, the user presses the end key combination. By default, this is #9.

7.15.3 Managing Call Recordings

Select "System > Recording." The Recordings page opens.



Figure 7-78. Recordings page.

The Recordings page enables you to select recordings by date for downloading and/or deleting. It displays the call time and the parties involved. The sample above shows two call recordings:

The first recording was made with the Record Calls option set to "Always."

The second recording was made with the Record Calls option set to "OnDemand."

- Start time: Date and time the recording was started.
- From: Extension number that made the call. On demand recordings are prefixed with OD.
- To: Extension that received the call.
- Action: Click "Download" to copy the recording to your computer. Click "Delete" to remove it.

7.16 Setting Up Fax Support

The Hybrid PBX and VoIP Gateway provides a fax passthrough feature that enables you to connect an analog fax machine to the analog phone network.

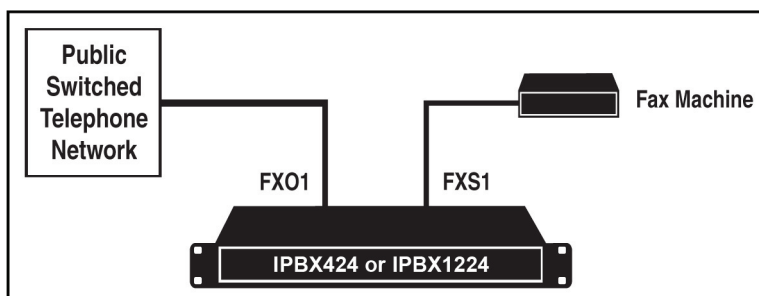


Figure 7-79. Fax setup.

To ensure that incoming and outgoing calls are properly routed to/from the fax machine, the following configuration settings need to be performed.

- The FXO port that the fax will connect to must be assigned to its own analog trunk.
- Incoming call rules for the fax trunk should point from the fax number to the extension assigned to the FXS port that the fax machine is plugged into.

Rule - Incoming-Fax

DID:

To: 6002 "Analog User"

CallerID Prefix Name:

CallerID Prefix Number:

Answer: ☐

Cancel

Save

Figure 7-80. Rule—Incoming Fax screen.

Rules - Incoming-Fax

RuleAction

All -> 6002

Edit

Delete

Up

Down

Back

Add

Figure 7-81. Rules—Incoming Fax screen.

8. Voicemail

8.1 Voicemail Configuration

Voicemail support is automatically enabled for each new user that you create. However, the voicemail service is disabled by default. It must be enabled and configured as described in the following sections.

8.1.1 Enabling Voicemail Support and Setting the Storage Location

1. Select "System > Maintenance > Logs" to open the Logs page.

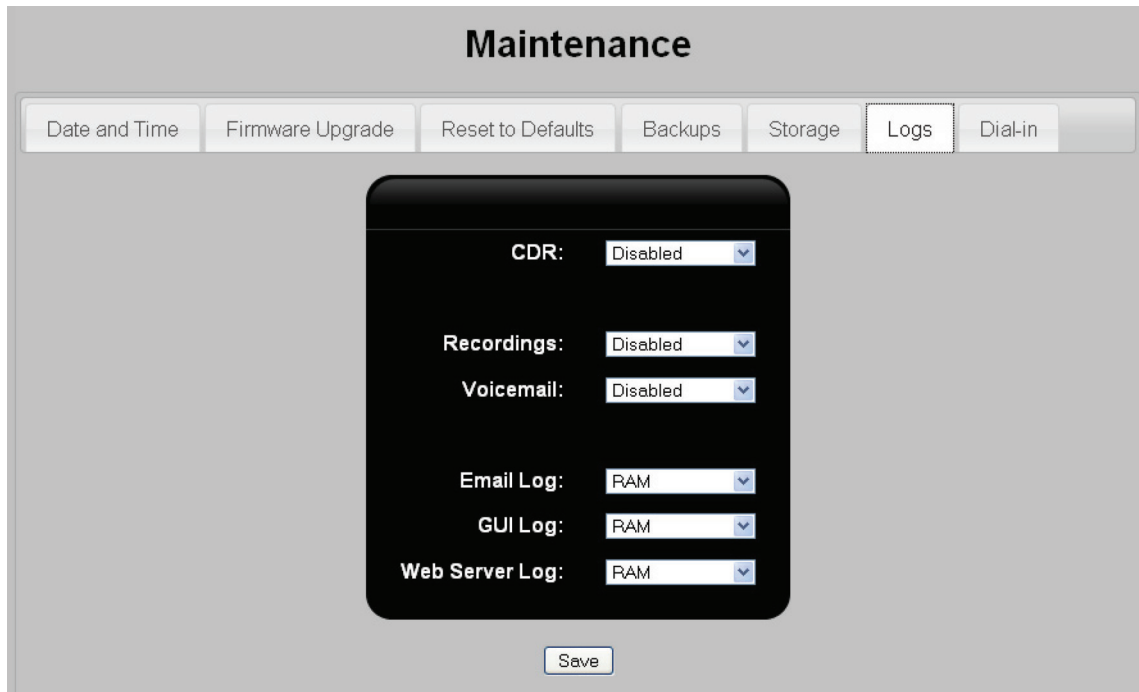


Figure 8-1. Logs page.

2. For Voicemail, select a storage option. This also enables voicemail support.

NOTE: You must restart the Hybrid PBX and VoIP Gateway for these settings to become active.

NOTE: If USB is specified, but not installed, then voicemail will be stored in system memory, and it will be lost when the system restarts.

3. Click "Save."

8.1.2 Configuring Global Voicemail Options

1. Select "PBX > PBX Settings > Voicemail." The Voicemail page opens.

The screenshot shows the 'PBX Settings' interface with the 'Voicemail' tab selected. The configuration options are as follows:

- Maximum Greeting Time: 60
- Minimum Message Time: 1
- Maximum Message Time: 60
- Maximum Messages: 15
- Exit on Key Press: ☒
- Transfer on '0' Key Press: 6000 "Sample IVR"
- Transfer on '*' Key Press: Outgoing Line Group "Sales"
- '*' Phone Number: (empty field)
- Allow Users to Review: ☒
- Say Message Caller ID: ☒
- Say Message Duration: ☒
- Send Messages by Email Only: ☐
- Attach Recording: ☐
- From: Voicemail
- Subject: [PTS-PBX]: New message \${VM_MSGNUM} in mailbox \${VM_MAILBOX} from \${VM_CALLERID}
- Body: Dear \${VM_NAME}:
just wanted to let you know you were just left a \${VM_DUR} long message (number \${VM_MSGNUM}) in mailbox \${VM_MAILBOX} from \${VM_CALLERID}, on \${VM_DATE}, so you might

A 'Save' button is located at the bottom of the form.

Figure 8-2. Voicemail page.

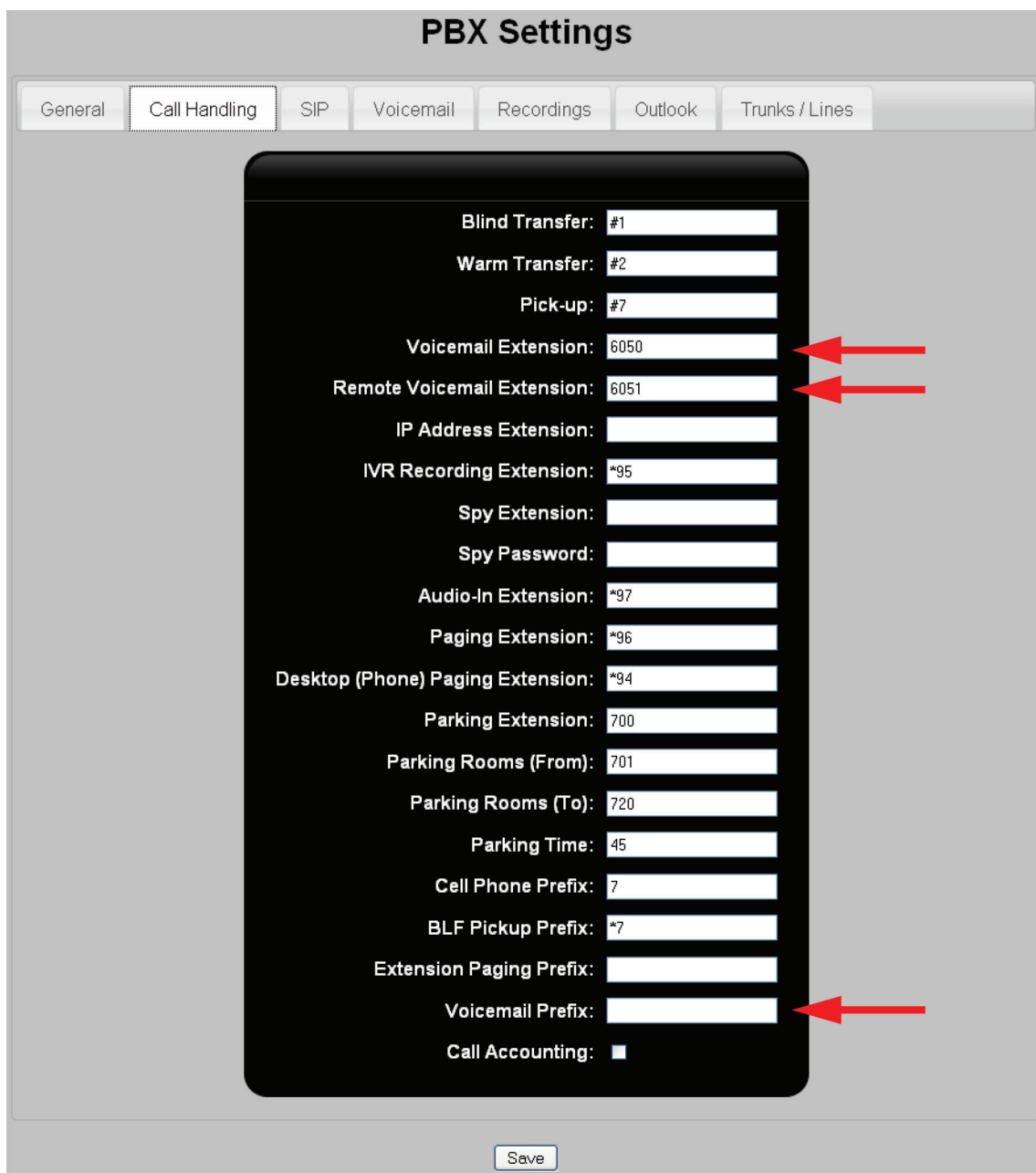
2. Configure the following settings:

- Maximum Greeting Time: Maximum time in seconds for user recorded voicemail greetings. Default: 60
- Minimum Message Time: Minimum length of time for a voicemail recording. Default: 1
- Maximum Message Time: Maximum length in seconds for a voicemail message. Default: 60
- Maximum Messages: Maximum number of messages allowed in a voicemail box. Default: 15
- Exit on Key Press: When enabled, activates the Transfer on "0" Key Press and Transfer on "*" Key Press options. These options can be useful if you want to allow quick access to the operator (or other extension), allowing callers to be answered immediately instead of leaving a message and waiting for a callback.
- Transfer on "0" Key Press: If a caller dials 0 while recording a voicemail, the caller is transferred to the selected extension, or to the configured external number which is automatically dialed on the selected outgoing line group.
- Transfer on "*" Key Press: If a caller dials * while recording a voicemail, the caller is transferred to the selected extension, or to the configured external number which is automatically dialed on the selected outgoing line group.
- Allow Users to Review: When enabled, callers are presented with the option to review and rerecord their voicemail message. Default: enabled
- Say Message Caller ID: When enabled, the caller's CallerID is added in audio to the voicemail message. Default: enabled
- Say Message Duration: When enabled, the message duration is added in audio to the voicemail message. Default: enabled
- Send Messages by Email Only: When enabled, voicemail messages are sent to the user's email address only, and are not saved on the Hybrid PBX and VoIP Gateway. Default: not enabled
- Attach recording: When enabled, voicemail messages are sent to the user's email address. When disabled, only a voicemail notification (without the voicemail) is sent. Default: enabled
- From: Specifies the contents of the "From" field in all emails that are sent. Default: Voicemail
- Subject: Specifies the content for the "Subject" field in all email messages that are sent.
- Body: Specifies the content for the body of all email messages that are sent.

3. Click "Save."

8.1.3 Configuring Voicemail Extensions and Prefixes

1. Select "PBX > PBX Settings > Call Handling."



The image shows a screenshot of the 'PBX Settings' interface, specifically the 'Call Handling' tab. The interface has a top navigation bar with tabs: General, Call Handling (selected), SIP, Voicemail, Recordings, Outlook, and Trunks / Lines. Below the tabs is a large black rounded rectangle containing various settings. Red arrows point to the 'Voicemail Extension' (6050), 'Remote Voicemail Extension' (6051), and 'Voicemail Prefix' fields. A 'Save' button is at the bottom.

Setting	Value
Blind Transfer:	#1
Warm Transfer:	#2
Pick-up:	#7
Voicemail Extension:	6050
Remote Voicemail Extension:	6051
IP Address Extension:	
IVR Recording Extension:	*95
Spy Extension:	
Spy Password:	
Audio-In Extension:	*97
Paging Extension:	*96
Desktop (Phone) Paging Extension:	*94
Parking Extension:	700
Parking Rooms (From):	701
Parking Rooms (To):	720
Parking Time:	45
Cell Phone Prefix:	7
BLF Pickup Prefix:	*7
Extension Paging Prefix:	
Voicemail Prefix:	
Call Accounting:	<input type="checkbox"/>

Save

Figure 8-3. Call handling screen.

2. Configure the following settings:

- Voicemail Extension: Extension to specify when users call in to check their personal voicemail messages. Default: 6050
- Remote Voicemail Extension: Extension to be used when users call in to check their voicemail messages from a different phone.

- Voicemail Prefix: Allows a message to be left in the extension's voicemail box without ringing the extension. If the value of *6 is entered then a call to *66011 would leave a message in the voicemail box of extension 6011 without ringing the extension. (*6 is only an example, there is no default value.)

3. Click "Save" and then click "Apply Configuration."

8.2 Setting up Voicemail to Email

The voicemail to email service allows voicemail recordings (or just notifications) to be sent to user email addresses.

8.2.1 Configure SMTP Settings

Configuration of the simple mail transfer protocol is required to support the voicemail to email feature.

1. Select "System > Network > SMTP." The SMTP page opens.

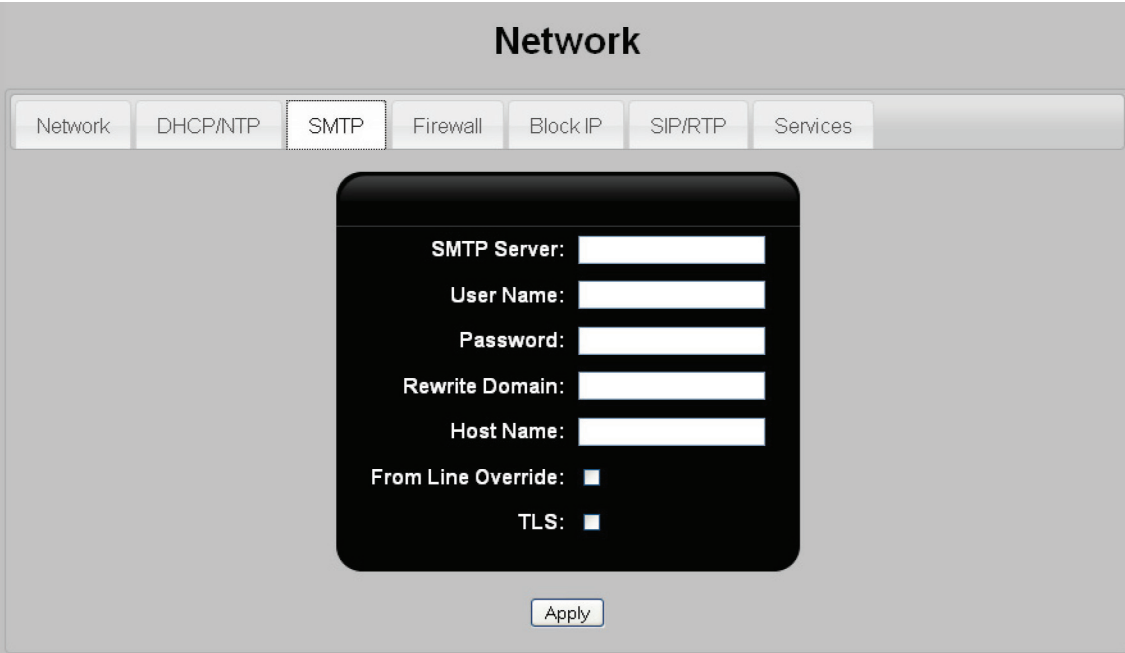


Figure 8-4. SMTP page.

2. Configure the following settings.

- SMTP Server: Specify the address of the mail server that the Hybrid PBX and VoIP Gateway will use to send emails. The address can include a colon followed by a port number if required.
- Username: Specify the user name for the email account that the Hybrid PBX and VoIP Gateway will use to send mail via the SMTP server. (May not be required by some servers.)
- Password: Specify the password for the email account that the Hybrid PBX and VoIP Gateway will use. (May not be required by some servers.)
- Rewrite Domain: Specify the text that will follow the @ symbol in the "From" address field for all emails sent by the Hybrid PBX and VoIP Gateway.
- Host Name: Specify the text that will precede the @ symbol in the "From" address field for all emails sent by the Hybrid PBX and VoIP Gateway.

- From Line Override: When enabled, overwrites the default value of the “From” header of an email. Default is set by selecting “PBX > PBX settings > voicemail.”
- Use TLS (Transport Layer Security): Enable this option if the SMTP server requires TLS.

3. Click “Save.”

4. Click “Apply Configuration.”

8.2.2 Configure Each User Account

To enable voicemail to email support for each user, you must configure the Email address in each user account.

- For SIP users, see Section 6.2.2.
- For analog users, see Section 6.2.3.
- For voicemail boxes, see Section 6.2.4.

You can also configure the Send Messages by Email Only option:

- For SIP users, see Section 6.2.2.
- For analog users, see Section 6.2.3.
- For voicemail boxes, see Section 6.2.4.

Finally, you can define this option globally for all users by selecting “PBX > PBX Settings > Voicemail” and setting the “Send Messages by Email Only” option. See Section 8.1.2.

8.2.3 Test Your Configuration

The following procedure explains how to test your voicemail configuration.

Assumption

- Two SIP phones, one on extension 6005, and the other on extension 6006, are connected/registered to the Hybrid PBX and VoIP Gateway.

To test voicemail

1. Pick up the phone at 6005 and dial “6006.”
2. When the phone at 6005 rings, do not answer the call.
3. Listen to the Unavailable message and then leave a voicemail message after the beep.
4. The Message Waiting Indicator (MWI) light on the 6005 phone should turn on.
5. Pick up the phone at 6005 and dial “6050,” or press the “envelope” button to check the voicemail.
6. The voicemail system prompts for the voicemail password of the user. The default voicemail password is “1234.”

8.3 Managing Voicemail

Users can manage voicemail by phone.

This section provides an overview of the menu options presented when a user retrieves voicemail. The digits in bold are the numbers the user presses to select a menu option. By default, a local user dials extension 6050 to retrieve voicemail. To customize this extension, see “Configuring voicemail extensions and prefixes” in Section 8.1.3).

Chapter 8: Voicemail

- 1 Listen to voicemail messages
 - 3 Advanced options
 - 1 Reply
 - Records message
 - 3 Envelope
 - 5 Send Message (only available if voicemail enabled)
 - Asks for extension to send to:
 - 4 Play previous message
 - 5 Repeat current message
 - 6 Play next message
 - 7 Delete current message
 - 8 Forward message to another mailbox
 - Asks for extension then:
 - 1 to prepend a message
 - 2 to forward without prepending
 - 9 Save message in a folder
 - 0 Save in New Messages
 - 1 Save in Old Messages
 - 2 Save in Work Messages
 - 3 Save in Family Messages
 - 4 Save in Friends Messages
 - * Help. During msg playback= Rewind
 - # Exit. During msg playback = Skip forward
- 2 Change folders
 - 0 Switch to New Messages
 - 1 Switch to Old Messages
 - 2 Switch to Work Messages
 - 3 Switch to Family Messages
 - 4 Switch to Friends Messages
- 3 Advanced Options
 - 5 Send Message (only available if voicemail enabled)
 - 1 Use Voicemail number (only available if voicemail enabled)
 - 2 Use Voicemail Directory (only available if voicemail enabled)
- 0 Mailbox options
 - 1 Record unavailable message
 - 2 Record busy message
 - 3 Record name
 - 4 Record temporary message (If there isn't already a temporary message) else:
 - 1 Record temporary message
 - 2 Erase temporary message (go back to the standard message)
 - 5 Change voicemail password
 - * Return to the main menu
- * Help
- # Exit

After recording a message (incoming message, busy/unavailable greeting, or name)

- 1 Accept
- 2 Review
- 3 Re-record

In all cases where a voicemail message is being played:

- * Rewind

- # Skip forward

9. Remote Office Setup

Setting up a connection to a remote office

If you have a Hybrid PBX and VoIP Gateway installed at another office, you can set up a connection to it to allow for the direct exchange of phone calls using extension numbers and a dialing prefix. In fact, each Hybrid PBX and VoIP Gateway PBX can be connected to multiple remote offices.

NOTE: For security reasons, we recommend that you make the connection to the remote office using a VPN (virtual private network) link.

To view all remote offices

Select “PBX > Remote Offices.” The Remote Offices list page opens.

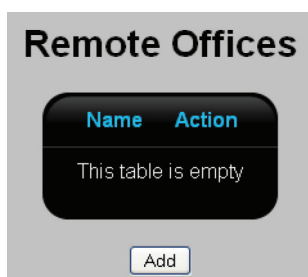


Figure 9-1. Remote offices.

This page lists all defined remote offices. The following information is presented for each office.

- Name: Name that identifies the remote office.
- Action: Click the appropriate button to Edit or Delete an office. The Edit button displays the Remote Offices edit page. See “Adding a remote office,” next.
- Add button: To add a new remote office, click Add, then see “Adding a remote office,” next.

Adding a remote office

1. Click “Add” on the Remote Offices list page, opening the Remote Office edit page.

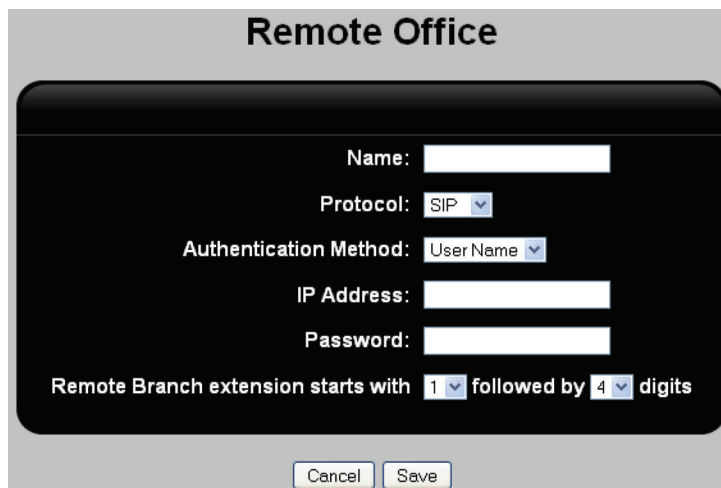


Figure 9-2. Remote office edit page.

2. Configure the following settings:

NOTE: To connect to a remote office, the Hybrid PBX and VoIP Gateway at each office must have the same settings for all parameters except IP address and Remote Branch extension starts with.

- Name: Specify a unique name to identify the connection. This must match the Name set on the remote office.
 - Protocol: Select the protocol to use on the connection: SIP or IAX2.
 - Authentication method: Select one of the following options:
 - User Name: Select this option if the two Hybrid PBX and VoIP Gateways are connected via the Internet. Make sure that Name and Password are identical on both units.
 - IP Address: Select this option if the two Hybrid PBX and VoIP Gateways are in the same office and connected via a LAN, or if they are in separate offices and connected using a VPN tunnel.
 - IP Address: Specify the IP address of the Hybrid PBX and VoIP Gateway WAN or LAN port at the remote office. If the port is hidden behind a router with a firewall or NAT, you need to specify the public IP address of the router, and then configure port forwarding on the router to pass SIP (or IAX2 traffic).
 - Password: Specify a password to protect the connection. The password should be at least twelve characters long and contain a combination of uppercase/lowercase letters, numbers, and at least one symbol. The same password must be configured at the remote office.
 - Remote Branch extension starts with X followed by Y digits: Select the prefix (X) that users will dial to reach the remote office, then select the number of digits used to identify extensions at the remote office (Y).
3. Click "Save" when you are done. The new office will be added to the Remote Offices list page.
4. Click "Apply Configuration" to activate the new remote office connection.

10. Maintenance

10.1 Setting the System Date and Time

Correct date and time settings are important for proper operation of the Hybrid PBX and VoIP Gateway. It ensures that voicemail and call data records are time-stamped correctly, and that the proper time is set on all SIP phones.

10.1.1 Automatically Setting the Date and Time

The most effective way to set the date and time is to use a time server. If the Hybrid PBX and VoIP Gateway is connected to the Internet, it can make use of a public time server to set the system time.

NOTE: All models have a clock with battery backup.

To configure access to a time server

1. Select "System -> Network -> DHCP/NTP." (If Enable DHCP Server is selected, see Configuring DHCP and NTP settings.)

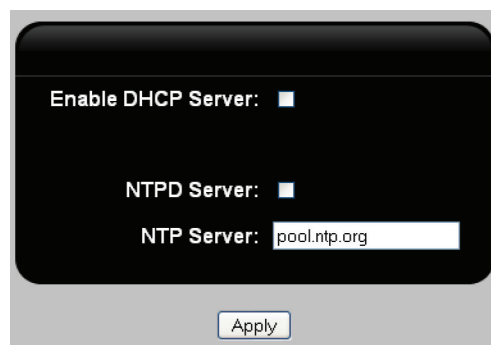


Figure 10-1. DHCP and NTPD screen.

2. Enable NTPD Server. When enabled, the Hybrid PBX and VoIP Gateway will set the system date and time based on values retrieved from the specified NTP Server.
3. Set NTP Server to the host name of a time server. The default setting is pool.ntp.org which is a public time server on the Internet.
4. Click "Apply."
5. Click "Apply Configuration."
6. For the retrieved data and time to be accurate you must configure the time zone in which the Hybrid PBX and VoIP Gateway is operating. See "Manually setting the date and time" in Section 10.1.2.

10.1.2 Manually Setting the Date and Time

1. Select "PBX > Maintenance > Date and Time."

Figure 10-2. Date and time screen.

2. Define date and time settings as described below. When done, click "Apply."
 - Time Zone: Select the time zone in which the Hybrid PBX and VoIP Gateway is located. When the Hybrid PBX and VoIP Gateway is connected to the Internet, it automatically retrieves accurate time and date settings (from an ntp server) based on the Time Zone you enter.
 - Current time: Shows the correct time on the Hybrid PBX and VoIP Gateway.
 - Year, Month, Day, Hours, Minutes, Seconds: Use these fields to define the new system time. By default, these fields are set to match the time and date on your computer.
3. Click "Apply."
4. Click "Apply Configuration."

10.2 Installing New Firmware

CAUTION: Before installing new firmware, it is important to do a system backup. See "Resetting the Hybrid PBX and VoIP Gateway" in Section 10.3.

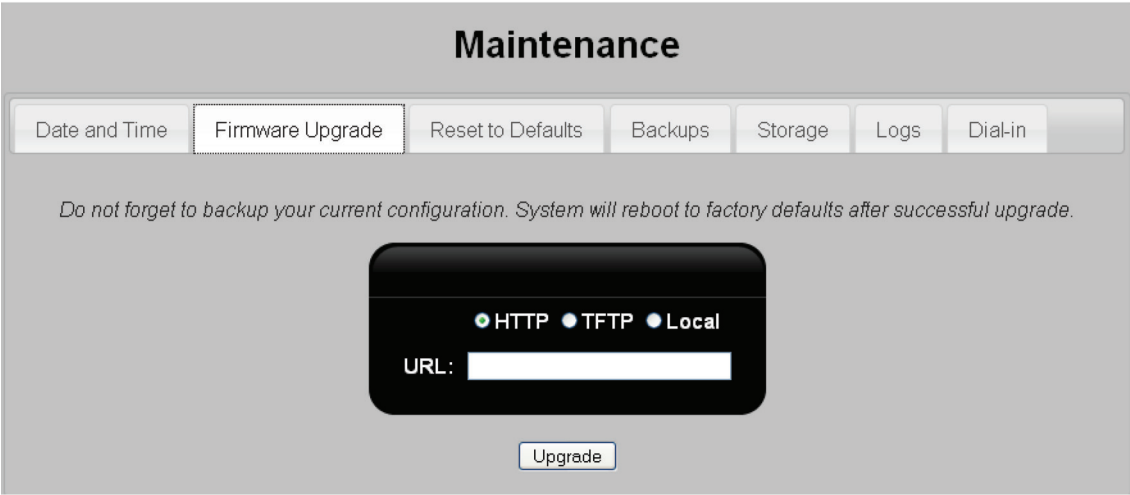
CAUTION: A firmware upgrade will disconnect all active calls.

10.2.1 Downloading the Latest Firmware

1. Point your Web browser to: <http://positrontelecom.com/supportD.html>.
2. Select the appropriate category for the model of Hybrid PBX and VoIP Gateway.
3. Locate and read the change-log.txt, version.txt pages before performing an upgrade.
4. Right-click the appropriate link and do a "Save Target As..." in your browser
5. Locate a directory for the download on your local file system.
6. Use the exact name specified and click "Save."
7. When using Internet Explorer the File name should be saved as "ulmage-md5" (with quotes), and the "Save as" type should be "All Files."

10.2.2 Installing the Firmware

1. Select "System > Maintenance > Firmware Upgrade."

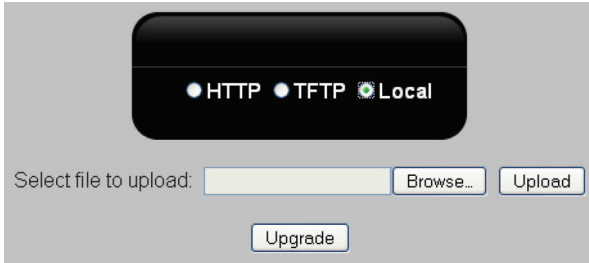


The screenshot shows the 'Maintenance' page with a tabbed interface. The 'Firmware Upgrade' tab is selected. Below the tabs, a warning message reads: 'Do not forget to backup your current configuration. System will reboot to factory defaults after successful upgrade.' The main area contains a black rounded rectangle with three radio buttons: 'HTTP' (selected), 'TFTP', and 'Local'. Below these is a 'URL:' label followed by a text input field. At the bottom of the black box is an 'Upgrade' button.

Figure 10-3. Firmware upgrade screen.

2. Choose and configure one of the installation options:

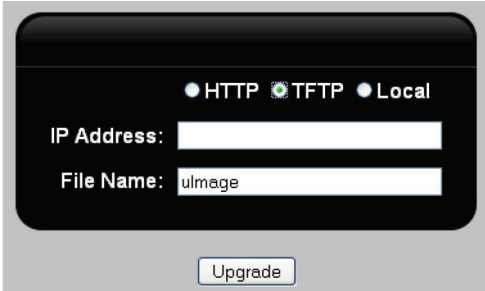
- Local: Click Browse to locate the firmware file, and then click Upload to load it onto the Hybrid PBX and VoIP Gateway.



The screenshot shows the 'Local' radio button selected. Below the radio buttons is a 'Select file to upload:' label followed by a text input field. To the right of the input field are 'Browse...' and 'Upload' buttons. At the bottom of the black box is an 'Upgrade' button.

Figure 10-4. Upload firmware file.

- TFTP: Specify the IP address of the TFTP server where the firmware file is located and the name of the firmware file.



The screenshot shows the 'TFTP' radio button selected. Below the radio buttons are two labels: 'IP Address:' followed by a text input field, and 'File Name:' followed by a text input field containing the text 'ulmage'. At the bottom of the black box is an 'Upgrade' button.

Figure 10-5. Specify the URL.

- HTTP: Specify the URL of the firmware file and click Upgrade. The Hybrid PBX and VoIP Gateway will retrieve the file and install it. For details, contact Black Box Technical Support at 724-746-5500 or info@blackbox.com.

3. Click "Upgrade."

4. A dialog box will indicate whether the upgrade succeeded or failed.

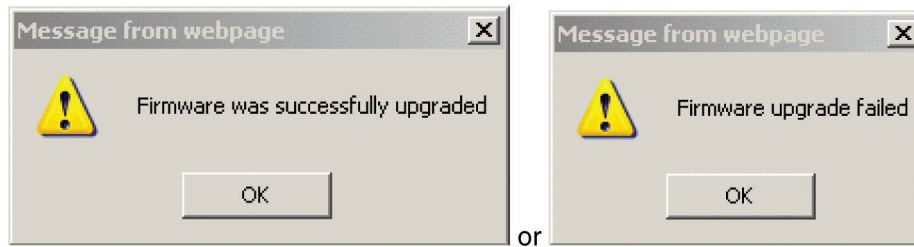


Figure 10-6. Upgrade result.

10.3 Resetting the Hybrid PBX and VoIP Gateway

To reset the Hybrid PBX and VoIP Gateway, select "System > Maintenance > Reset to Defaults."

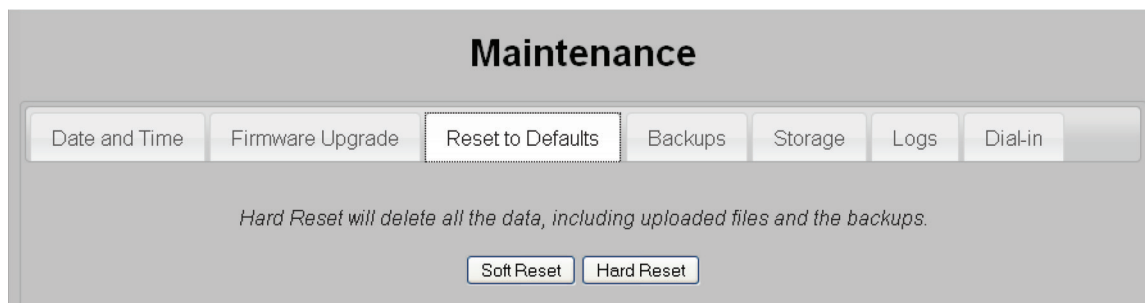


Figure 10-7. Reset to defaults.

Two options are available: soft reset and hard reset. Both options restore the Hybrid PBX and VoIP Gateway to its original factory settings, deleting all configuration settings. A soft reset preserves backups, while a hard reset deletes them.

1. Click either button to start the reset procedure.
2. You will be prompted to continue. Click "OK" to proceed.

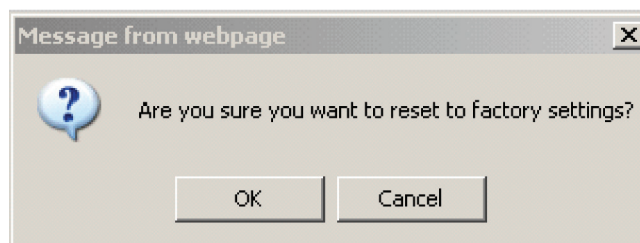


Figure 10-8. Continue prompt.

3. During the reset a countdown timer will be displayed. When the timer expires the reset should be complete and you will be able to log into the command center again.

10.4 Restarting the Hybrid PBX and VoIP Gateway

To restart the Hybrid PBX and VoIP Gateway, select System > Restart. After the restart, existing configuration and backup files remain intact, however any data stored in system memory is deleted.

When a restart is performed, any telephone calls in progress are dropped and any logs in system memory are cleared.

During the restart a countdown timer will be displayed. When the timer expires the restart should be complete and you will be able to log into the command center again.

10.5 Backing Up and Restoring Configuration Settings

The Hybrid PBX and VoIP Gateway has a backup and restore function that enables you to save and restore configuration settings.

- Backups are stored in system memory, but can also be downloaded as a tar.gz file.
- Backups do not interfere with system operation and can be performed at any time.

To perform a backup or restore, select “System > Maintenance > Backups.”

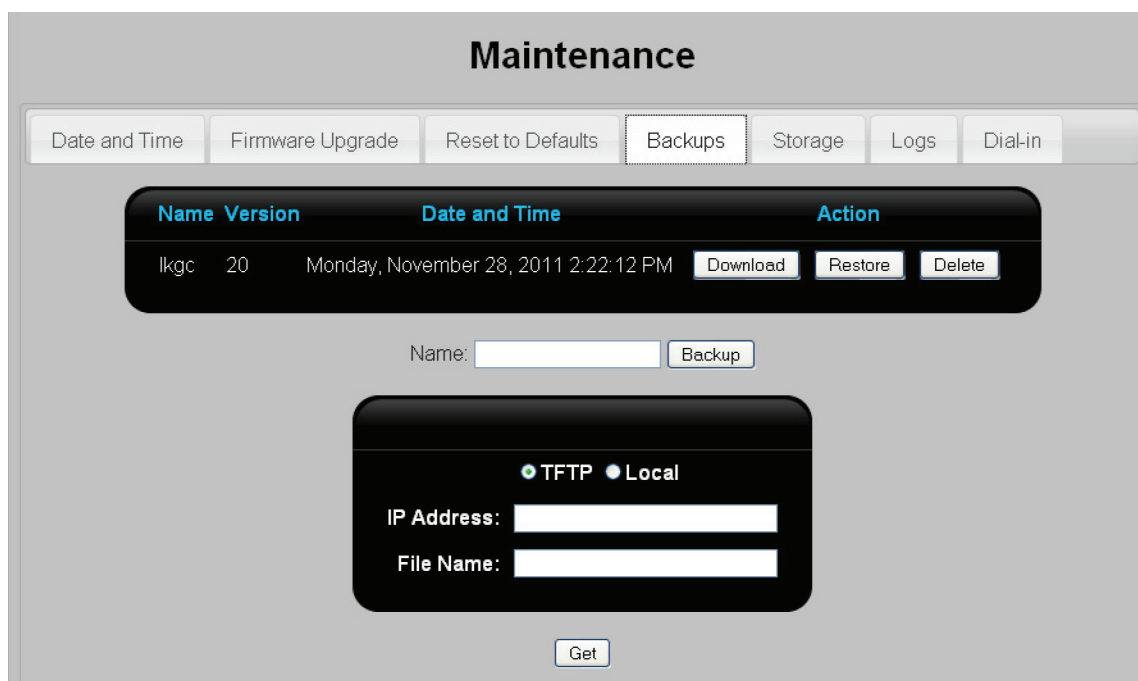


Figure 10-9. Backups screen.

The table at the top of the page displays all backups that are stored on the system.

Each time you click “Apply Configuration” in command center, the backup file lkgc (last known good configuration) is updated.

- Download: Copies the file to your computer.
- Restore: Updates the current configuration with settings from the backup file.
- Delete: Removes the backup file.

10.5.1 Creating a Backup

Specify a name for the backup in the Name box, then click “Backup.”

10.5.2 Restoring a Backup

You can restore a backup directly from the backup list by clicking the “Restore” button. Or, you can use the TFTP or Local options to load a backup file from another location.

- TFTP: Specify the IP address of the TFTP server where the firmware file is located and the name of the firmware file.

Click “Restore” next to the file you want to restore.

1. Select “System > Maintenance > Backups.”

Name	Version	Date and Time	Action
lkgc	20	Monday, November 28, 2011 2:22:12 PM	<button>Download</button> <button>Restore</button> <button>Delete</button>

Name: Backup

☒ TFTP ☐ Local

IP Address:

File Name:

Get

Figure 10-10. Backups screen.

2. Choose and configure one of the backup options:

- Local: Click “Browse” to locate the backup file, select it, and then click “Upload.”

☐ TFTP ☒ Local

Select file to upload: Browse... Upload

Figure 10-11. Choose a backup option.

- TFTP: Specify the IP address of the TFTP server where the backup file is located and its name, then click “Get.”

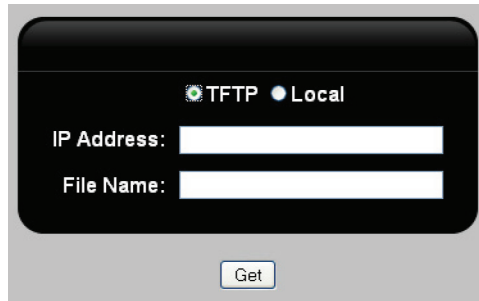


Figure 10-12. Specify IP address.

3. A dialog box will be displayed if the backup was successful.

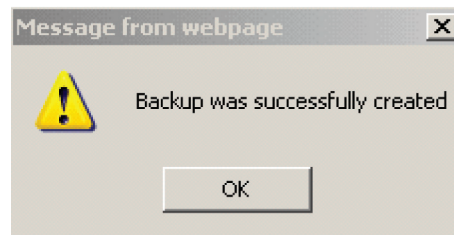


Figure 10-13. Backup successful.

10.6 Formatting Auxiliary Memory

The Hybrid PBX and VoIP Gateway uses auxiliary memory to store:

- Voicemail for each user.
- Voicemail greetings.
- Call recordings.
- Log files. See “Log files” in Section 10.9.

If auxiliary memory is not installed, these items are stored in system memory. A limited amount of system memory is available for this purpose, and once full, older items are discarded and replaced with newer entries. We strongly recommend installing auxiliary memory.

NOTE: One GB of auxiliary memory will store approximately 1000 minutes of voicemail and voice recordings.

- Auxiliary memory must be formatted by the Hybrid PBX and VoIP Gateway before it can be used. See “Formatting a USB flash drive” in Section 10.6.2.
- Auxiliary memory devices larger than 50 GB must be formatted on a computer before installation. See “Formatting auxiliary memory devices on a computer” in Section 10.6.3.

USB flash drive

Although the Hybrid PBX and VoIP Gateway has two USB ports, only one USB flash drive can be installed at a time. The USB drives can be installed while the Hybrid PBX and VoIP Gateway is running.

CAUTION: Never remove a USB flash drive when the Hybrid PBX and VoIP Gateway is on. This can result in a loss of data.

10.6.1 Formatting a USB Flash Drive

1. Select "System > Maintenance > Storage."

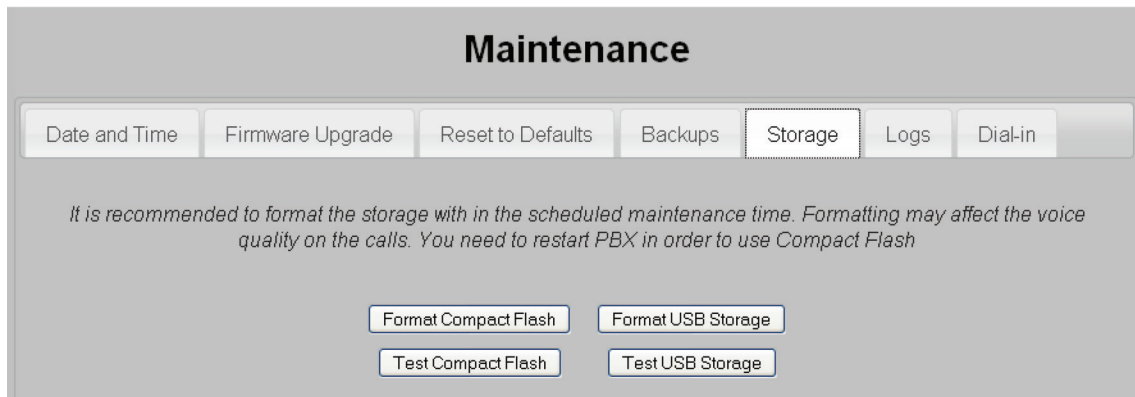


Figure 10-14. Storage screen.

2. Remove any USB device (such as a 3G data adapter) that is also installed.
3. Click "Format USB Storage."
 - The estimated time for a USB format is approximately 1.5 minutes/GB.
 - All data on the USB flash drive will be lost when it is formatted.
4. If error messages are displayed during formatting, select "Diagnostics > Logs > GUI" for details.
5. When formatting is complete, remove and re-install the USB flash drive.
6. Test the USB flash drive by clicking "Test USB Storage." A message will appear to indicate the success or failure of the test.
7. Summary information is presented on the Status page. For example:

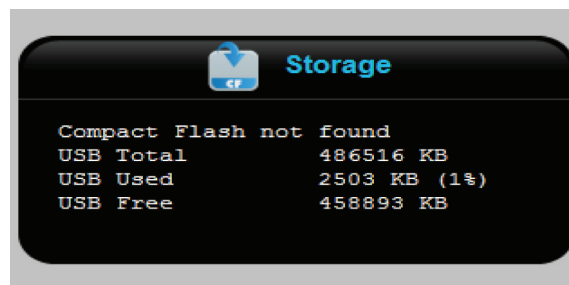


Figure 10-15. Status page.

8. Re-install any USB device (such as a 3G data adapter) that was removed.

10.6.2 Formatting Auxiliary Memory Devices on a Computer

Auxiliary memory devices larger than 50 GB must be formatted on a computer before installation. The required file format is ext2, which can be created using third-party software such as the Gnome Partition Editor—GParted Live (available for download at: <http://gparted.sourceforge.net/index.php>).

10.7 Installing a Language Pack

You can install a language pack to provide support for additional languages. By default, the English language pack is installed.

To see the list of supported languages, select “PBX > Sound Manager > Language.”



Figure 10-16. Language page.

If you want to load a new version of an installed language, first delete it, then download and install the new version.

NOTE: To install a new language pack, contact Black Box Technical Support at 724-746-5500 or info@blackbox.com.

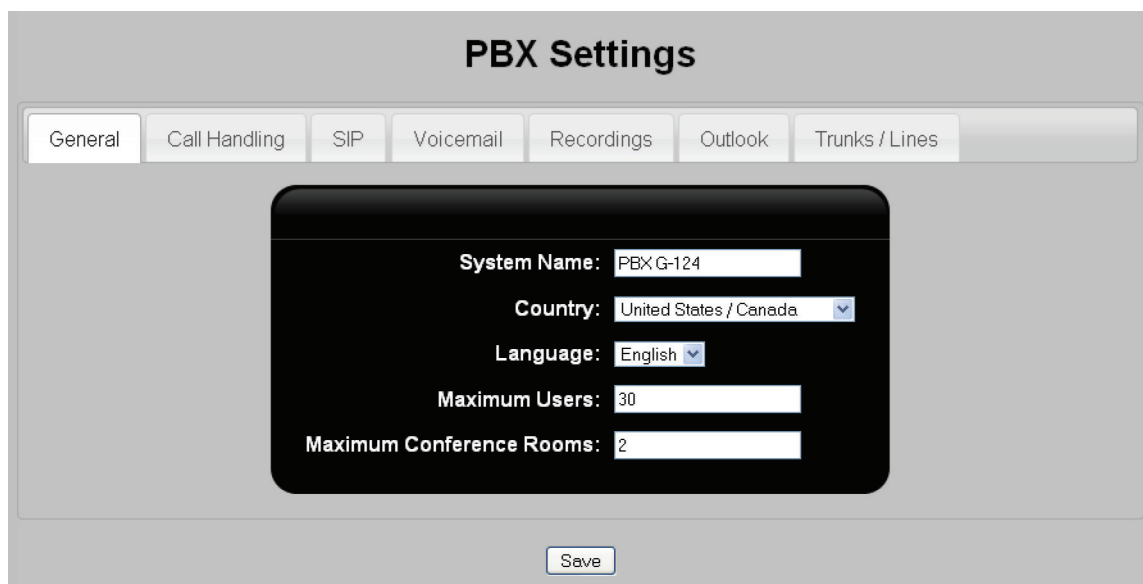


Figure 10-17. General settings.

6. For Language, select the new language you just installed.
7. Click "Save."
8. Click "Apply Configuration." The new language will now be active.

10.8 Diagnostic Tools

Select "Diagnostics > Diagnostics" to reach the following three pages.

10.8.1 Ping

The Ping command offers a way to test the connection between the Hybrid PBX and VoIP Gateway and a device connected to the WAN or LAN ports. For example, you can use Ping to verify if the connection to the Internet is working, or if a particular phone or VoIP provider is reachable.

To use Ping:

1. Select "Diagnostics > Diagnostics > Ping."

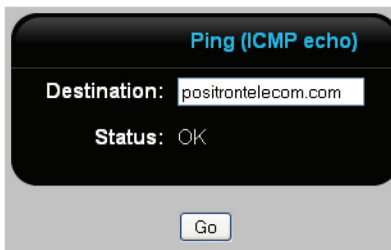


Figure 10-18. Ping screen.

2. For Destination, specify the IP address or URL of the target. For example:
 - To test your Internet connection, specify the IP address or URL of a site on the Internet. For example: blackbox.com.
 - To test the connection to a phone, specify the phone's IP address.
3. Click "Go."
 - If Status displays "OK," then the connection is working.
 - If Status displays "No response," then select "System > Network > Network" and make sure that the Default Gateway and DNS server are set to the correct values.

10.8.2 Console

The Console page lets you isolate PBX messages following a particular activity. This page will generally be used in connection with Black Box technical support to diagnose configuration issues.

To capture PBX messages for a particular issue:

1. Click "Start."
2. Perform the action on the PBX or on a phone that is causing the issue.
3. Click "Stop." Any logged messages will be displayed.
4. Click "Download" to save the file to your computer so that you can email it to Black Box for analysis.

10.9 Log Files

The Hybrid PBX and VoIP Gateway provides several log files that can be useful for viewing status information and when troubleshooting technical issues.

To enable the log files

Select "System > Maintenance > Logs" to open the Logs page. This page lets you define which logs will be enabled and where the log file data will be saved.

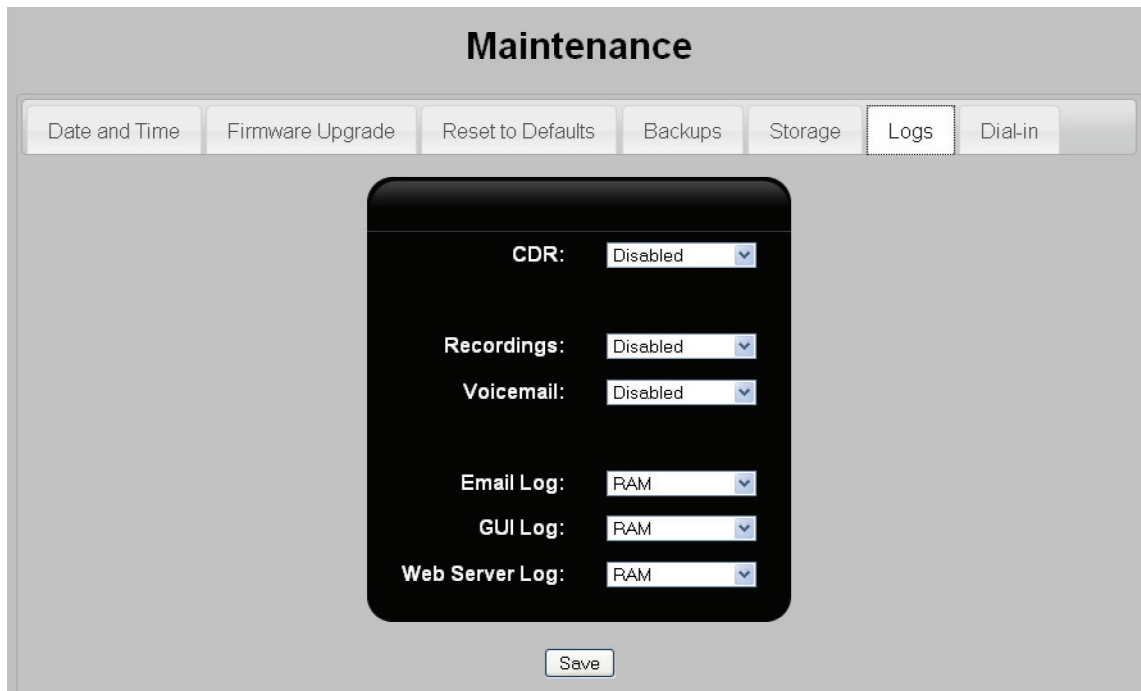


Figure 10-19. Logs screen.

Logging options

The following configuration options are available for each log file:

- Disabled: The log is off.

- USB: Log data is stored on the USB flash drive.
- RAM: Stores data in RAM. Data in RAM is cleared each time the unit is restarted. The RAM available for logs is limited. When all available memory is used, the log overwrites itself.

NOTE: If USB storage is specified, but not installed, then the associated log file will be stored in system memory, and log data will be lost when the system restarts.

Log file list

- CDR: This controls the call detail record. For details, see “Tracking call information” in Section 7.13.
- Recordings: This controls call recordings. For details, see “Call recordings” in Section 7.15.
- Voicemail: This controls voicemail. For details, see “Chapter 8: Voicemail.”
- Email Log: Log of all messages that have been sent by the Hybrid PBX and VoIP Gateway. The log is useful for troubleshooting email-related problems. View this log on the “Diagnostics > Logs” page.
- GUI Log: Contains all messages related to the command center web interface. View this log on the “Diagnostics > Logs” page.
- Web Server Log: Contains a log of the requested pages and files from the internal web server—useful in troubleshooting problems with file serving for phone provisioning. View this log on the “Diagnostics > Logs” page.

10.10 Accessing the Command Center over an Analog Dial-Up Connection

The dial-in service enables you to access the command center on the Hybrid PBX and VoIP Gateway from a remote site by using a dial-up connection over an analog phone line.

10.10.1 Configuring Dial-In Support

Before you can use one of the dial-in scenarios described later in this section, you need to configure dial-in support. Follow these instructions:

1. Select System > Maintenance > Dial-In.

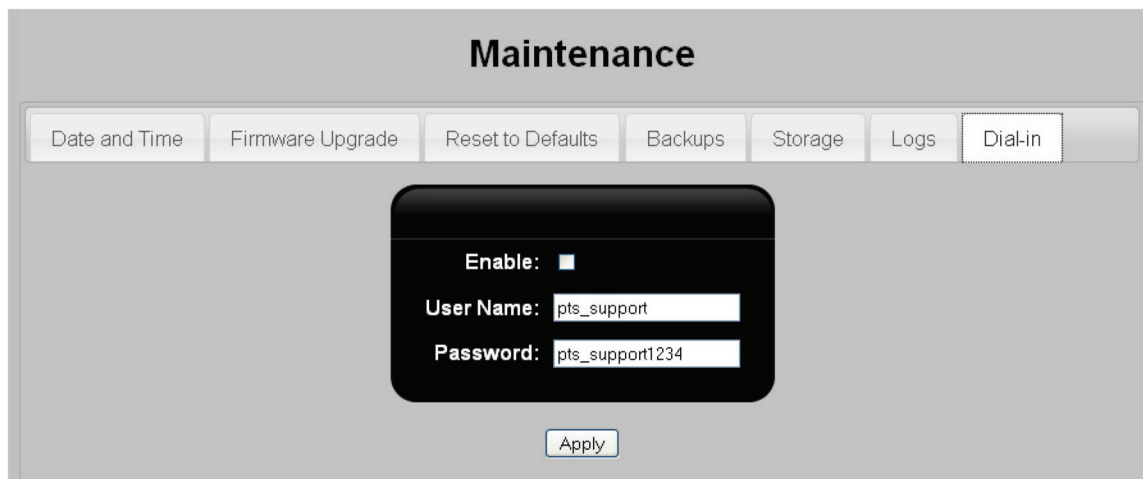


Figure 10-20. Dial-In screen.

2. Select "Enable."
3. For User Name, specify the username that must be provided when dialing in.
4. For Password, specify the password that must be provided when dialing in.

10.10.2 Dial-In Scenarios

There are two ways to setup dial-in support: dedicated line or shared line.

Dedicated line

This is the easiest way to setup dial-in support. The Hybrid PBX and VoIP Gateway is connected to an analog phone line via a USB modem connected to its USB port. The line is dedicated to dial-up only, and cannot be used to provide analog calling for PBX users.

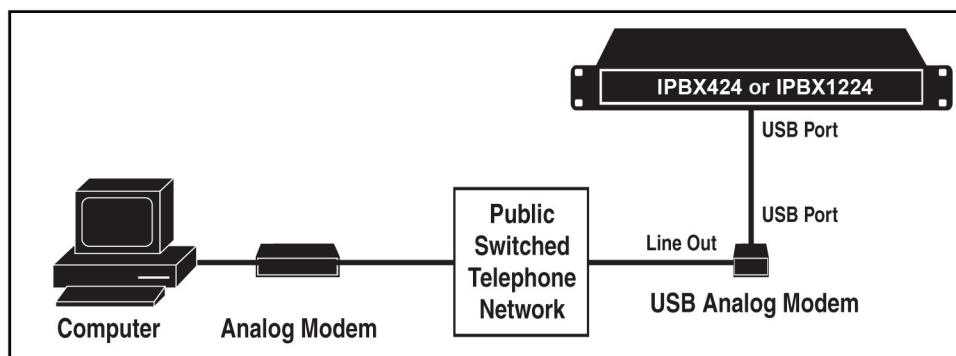


Figure 10-21. Dedicated line used for dial-in.

Configuration steps

1. Plug a USB modem into one of the USB ports on the Hybrid PBX and VoIP Gateway.

NOTE: The only supported modem is the USB RoboticsUSR5637.

2. Connect the line out on the modem to an analog phone line.
3. Setup a dial-up connection on the remote computer with the following values:
 - Set the user name and password you configured for dial-in support in the previous section.
 - Set the number to dial as the number assigned to your FXO1 line. (This example uses FXO1, but you can use another FXO port instead.)

For example, in Windows, a dial-in connection configuration would look like this:

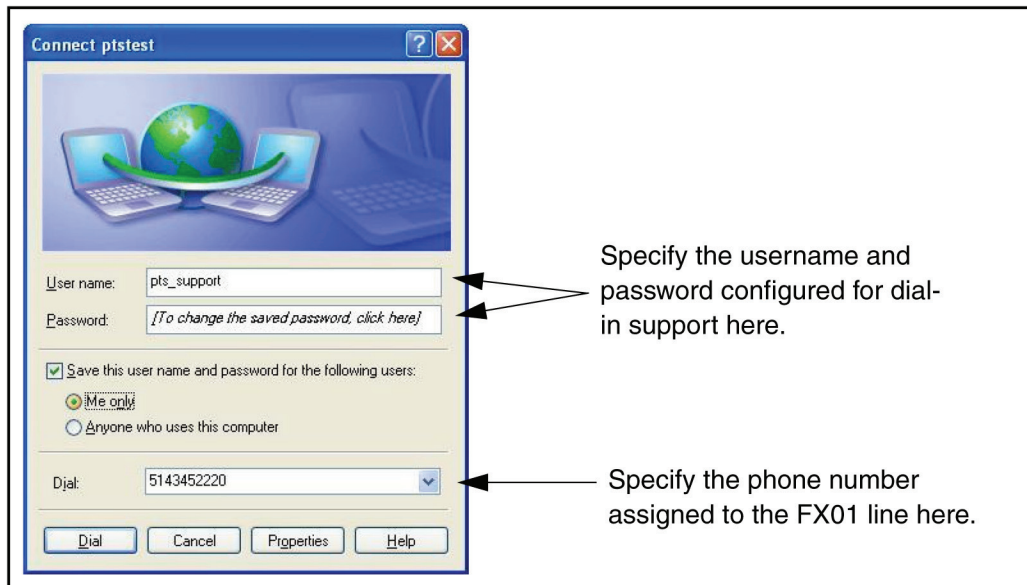


Figure 10-22. Dial-in connection configuration.

4. Start the connection. (In this example, you would click "Dial.")
5. Once the call is connected, open your Web browser and enter the address 192.168.200.1 into the line address bar and press "Enter." The Hybrid PBX and VoIP Gateway command center should start.

Shared line

In this scenario, the Hybrid PBX and VoIP Gateway is connected to an analog phone line via an FXO port. (FXO1 is the port used in the diagram, but any other FXO port could be used.) This allows the phone line to be used for analog calls as well as dial-in.

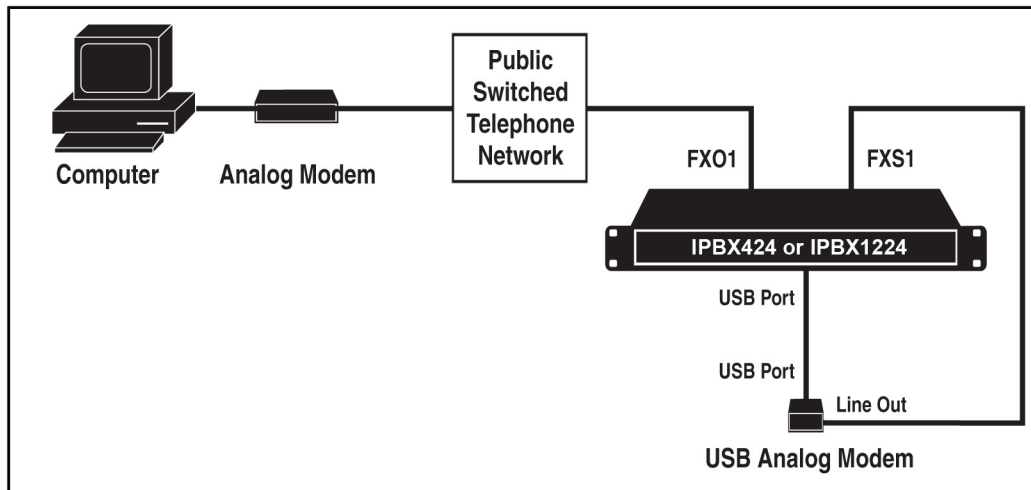


Figure 10-23. Shared line configuration.

Dial-in calls must be handled via a USB port. To complete the setup, the USB modem is connected to the USB port on the Hybrid PBX and VoIP Gateway, and the USB line out is connected to an FXS port. (FXS1 is used in the diagram, but any other FXS port could be used.)

Next, an incoming call rule and IVR are set up to allow the dial-in call to reach the USB port via the FXS port and the USB modem.

Configuration steps

1. Plug a USB modem into one of the USB ports on the Hybrid PBX and VoIP Gateway.

NOTE: The only supported modem is the USB Robotics USR5637.

2. Connect the line out on the modem to FXS1. (This example uses FXS1, but you can use any FXS port instead.)
3. Connect FXO1 to an analog phone line. (This example uses FXO1, but you can use any FXO port instead.)
4. Setup a dial-up connection on the remote computer with the following values:

- Set the user name and password you configured for dial-in support in the previous section.
- Set the number to dial as the number assigned to your FXO line. Follow this with the approximate command characters to insert a pause (commas in this example), then add the extension number of the FXS line. The pause is required to wait for the welcome message to play when the IVR answers the call, before dialing the extension number. In the example below, the number 51434562220 is followed by a 5-second pause, and then the extension 6002 is dialed to reach the Hybrid PBX and VoIP Gateway PBX command center via FXS1.

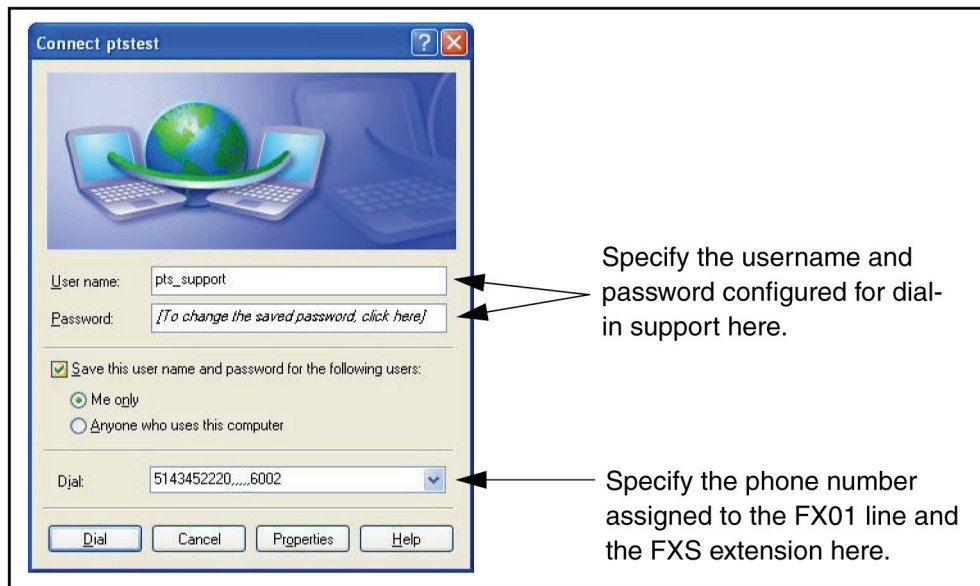


Figure 10-24. Setting up a dial-up connection.

5. Start the connection. (In this example, you would click "Dial.")
6. Once the call is connected, open your Web browser and enter the address 192.168.200.1 into the address bar and press "Enter." The Hybrid PBX and VoIP Gateway command center should start.

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