

RedPort

GMN
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Administrator's Guide

RedPort VoIP Routers

wXa-400; wXa-404; wXa-408

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GMN RedPort VoIP Administrator Guide Revision History

Date	Revision	Author
18 May 2012	Revision 1: Initial release	Susan Beck

1 About this guide

1.1 Who Should Use this Guide

This guide is intended for administrators of the RedPort VoIP system.

wXa refers to the webXaccelerator, a trademark of Global Marine Networks, LLC.

VoIP is the Voice over Internet Protocol system.

Administrators will learn how to:

- Connect and access their VoIP system
- Activate the VoIP system
- Make a call
- Customize the VoIP configuration
- Check the VoIP system status
- Add extensions
- Create user accounts
- Route incoming calls to different destinations
- Access Call Detail Records (CDRs)
- Set up Interactive Voice Response (IVR)
- Change recordings
- Use the database manager
- Backup their system

The following chapter references will help you in configuring your RedPort VoIP system whether you are performing an initial configuration or implementing one of the more advanced configuration schemes. Note these chapter references are internal links for this Administrator Guide.

1.1.1 Initial Configuration

For initial configuration, refer to the following chapters:

[Introduction](#)

[Setting Up Your RedPort VoIP system](#)

[Connecting to wXa Router Web Administrator](#) (wXa-503 Model only)

[Connecting to VoIP Web Administrator](#)

[Starting your VoIP System](#)

[Making a Call](#)

[Configuring Extensions](#)

[User Accounts](#)

[Call Detail Records \(CDRs\)](#)

[Pincode Records](#)

[Backing up your System](#)

1.1.2 Advanced Configurations

Find information for more advanced configurations in the following chapters:

[The IVR \(Interactive Voice Response\) Menu](#)

[Changing your Recordings](#)

[Using the Database Manager](#)

1.2 Conventions

This document uses the following typographical conventions:

1. Command and option names appear in bold type in definitions and examples.
2. Variable information appears in italic type.
3. Screen output, examples and code samples appear in `courier` type.

In addition, the following symbols may be used in command syntax definitions.

4. Square brackets [] surround optional items.
5. Angle brackets < > surround user-supplied values.
6. Pipe symbol | separates mutually exclusive values for an argument.

Note: General notes will appear like this.

Comments will appear in a box like this.

Warning: Warnings will appear like this.

Web links will look like this: www.globalmarinenet.com
www.redportglobal.com

2 Introduction

Global Marine Networks (GMN), the leaders in advancing satellite data speeds and services, helps Fixed and Mobile Satellite Services providers and their customers by offering the industry's fastest, most reliable and easy-to-use email, web, VoIP and other hardware and software services to maritime, oil and gas, first responder and business continuity users. The company's products include XGate high-speed satellite email, WeatherNet weather and oceanographic data software, and vessel tracking systems.

Ship to shore network management solutions are sold by GMN under the RedPort Global brand name at www.redportglobal.com and as white-label solutions for the world's premier satellite data service providers.

GMN has numerous awards and certifications for technical innovation and holds pending patents on its products. For more information on how GMN is Making Airtime Count™ - whether ship to shore, or in remote or emergency communications environments visit www.globalmarinenet.com.

2.1 RedPort VoIP system features

RedPort VoIP is a full featured industrial VoIP appliance which is optimized for use over satellite links. It is designed to produce exceptional voice quality while consuming very little bandwidth. With VoIP, owners and crew are able to make and receive calls using VoIP services with standard phone numbers based in 30 different countries. Land based callers are able to take advantage of local phone numbers while also keeping costs low.

RedPort VoIP Core Features include:

- Ship business lines with phone number selection available from 30 different countries
- Pincode based prepaid calling for crew and passengers
- Support of up to 8 analog phone ports (dependent upon your specific VoIP system)
- Support for unlimited VoIP/SIP Internet phones including WiFi phones
- Support for software based phones (i.e. soft phones)
- Up to 25 minutes of talk time per Mbyte of data consumed; Average system usage is 6kbps of bandwidth per call

- Up to 8 concurrent calls over one satellite link
- Inexpensive termination rates
- Web based graphical interface allowing for remote office support
- A fully configurable and extensible PBX with the following features:
 - IVR Menu for customization of inbound call recording
 - Call Detail Records
 - Contact Manager
 - Extension configuration
 - Voice Mail accessibility from web and email
 - Announcements
 - Call Broadcast
 - Call Forward
 - Call Monitoring
 - Call Parking
 - Call Pickup
 - Call Queuing
 - Call Recordings
 - Call Routing
 - Call Transfer
 - Call Waiting
 - Caller ID
 - Conferencing
 - Dial by Name
 - Do Not Disturb (DND)
 - Follow-me feature
 - Customization of Inbound Routes and Outbound Routes
 - Intercom
 - Music On Hold
 - Paging (dependent upon phone model)
 - Provisioning (dependent upon phone model)
 - Hunt Groups/Ring Groups
 - Routing by Caller ID
 - Text-to-Speech customization
 - Three-way Calling

2.2 Applicable Systems

This guide is designed for administration of VoIP system models **wXa-400**, **wXa-404**, **wXa-408**, and **wXa-503**. wXa-503 is a Router and VoIP system.



Figure 2-1 wXa-503 wXa Router and VoIP System

Model	Features
wXa-400	VoIP System; 0 analog ports
wXa-404	VoIP System; 4 analog ports
wXa-408	VoIP System; 8 analog ports
wXa-503	wXa Router and VoIP system

2.3 Manufacturer Contact Information

RedPort VoIP Systems are manufactured by:

Global Marine Networks, LLC

2668 Jericho Rd,

Maryville, TN 37803

Tel: +1.865.379.8723

Fax: +1.865.681.5017

Web: www.globalmarinenet.com

Email : info@globalmarinenet.com

Internal components manufactured in Germany, Taiwan, and USA.

Assembled in the Netherlands.

2.4 Warranty

RedPort VoIP is warranted for a period of 12 months. If the product malfunctions within 12 months of purchase the product is eligible for RMA (Return Material Authorization) replacement or repair at the manufacturer's discretion. No units will be accepted without an RMA. Units received without an RMA number will be returned unopened.

This warranty does not cover:

- Units damaged by environmental conditions such as operating in excessively humid, hot, cold, or salty environments
- Direct exposure to water, sunlight, or elements
- Reverse electrical polarity or operating with improper voltage or current
- Physical damage to the case or any of its internal components
- Inappropriate electrical connections to any of the devices physical ports
- Exposure to strong microwave or other EMF radiation such as that caused by lightning or other radio transmitting equipment
- Any other environmental, electrical, or mechanical exposure deemed inappropriate by the manufacturer

2.4.1 RMA Request

To request an RMA send an email to support@globalmarinenet.com with the following information:

- Name and address
- Original order number and/or copy of the order receipt
- Description of the problem

2.4.2 Shipping

Shipping costs and insurance from origin to Global Marine Networks is the responsibility of the end user. Global Marine Networks will cover return costs via UPS Ground to any USA location. Customers are responsible for covering international shipping costs.

Items with RMA approval should be shipped to the manufacturers address listed in section 2.3.

The RMA number should be clearly marked on the outside of the package, the packing slip, and shipping label.

3 Setting Up Your RedPort VoIP system

3.1 Inventory

The following parts are **included** with the purchase of your RedPort VoIP system:

- RedPort VoIP appliance (Note: wXa-503 also includes RedPort wXa router)
- US or EU compatible external power supply

The following parts are **not included** and will need to be purchased separately from your dealer or a supply store, if desired:

- Analog phone(s)
- Ethernet phone(s): SNOM brand Ethernet phones are recommended and installation instructions for the SNOM phone can be found in [Configuring a SNOM Phone](#)
- Laptop or desktop computer system for administration and configuration and for use as a console
- Ethernet cables

3.2 Quick Start

Quick start guides are included in Appendix A of this manual and on the DVD included with your purchase of the RedPort VoIP system. Links to the Quick start guides in Appendix A are included here for convenience:

- [Quick Start Guide for VoIP System](#)
- [Quick Start Guide for wXa](#)

3.3 Connecting your System

3.3.1 Models wXa-400, wXa-404, wXa-408

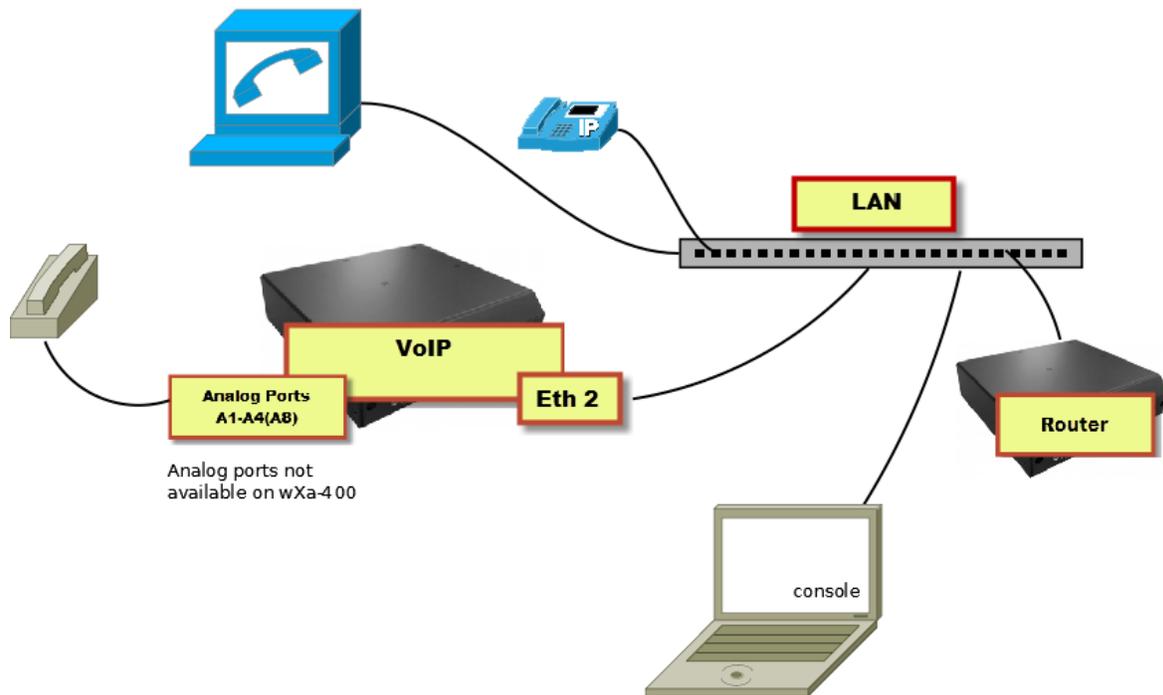


Figure 3-1 Model wXa-4xx Connections

Model wXa-4xx Ports	Purpose
A1-A4 (A8)	Connect to Analog Phones
ETH2	Connect this port to your Local Area Network (LAN)

3.3.2 Model wXa-503

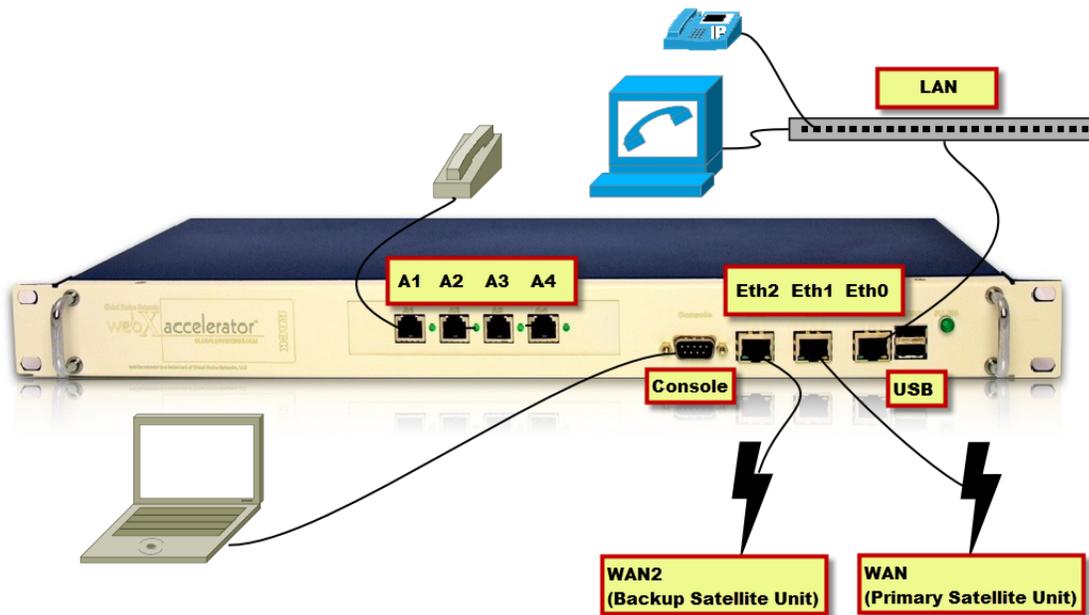


Figure 3-2 wXa-503 Connections

Model wXa-503 Ports	Purpose
A1-A4	Connect to Analog Phones
Console Port	Connect to laptop or PC with DB-9 NULL modem cable (Note: USB adapter may be needed for your laptop or PC)
ETH0	Connect this port to your Local Area Network (LAN). IP Phones and Soft phones are connected via your LAN as depicted in the diagram.
ETH1	Connect this port to your Primary Satellite Unit
ETH2	Connect this port to your Backup Satellite Unit (if applicable)

4 Connecting to wXa Router Web Administrator

This section applies to model number wXa-503 only.

After setting up your system following the connection diagram in [Connecting your System](#), start a web browser and enter URL **http://<IP address of wXa LAN >** to connect to the **wXa Router Web Administrator**.

Note: IP address 192.168.10.1 is the default IP address for the wXa LAN.

The login for the wXa Router Web Administrator is

```
Username: admin
Password: webxaccess
```

The wXa Router Web Administrator opens:

The screenshot displays the webX accelerator web administrator interface. At the top, it features the logo for Global Marine Networks and the product name 'webX accelerator'. The URL 'webxaccelerator.gmn-usa.com' is visible in the top right corner. A navigation menu includes 'System', 'Interfaces', 'Firewall', 'Services', 'VPN', 'Status', and 'Diagnostics'. The main content area is titled 'System Overview' and contains a table of system information.

System information	
Name	webxaccelerator.gmn-usa.com
Version	1.2.3.23x-RELEASE built on Mon Mar 12 22:33:29 UTC 2012
Run Status	Operational
Uptime	00:17
Default (WAN)	308299/76146 (301 KB/74 KB) Change default route/interface
State table size	26/10000 Show states
MBUF Usage	261 /645
CPU usage	 1%
Memory usage	 14%
Disk usage	 27%

At the bottom of the interface, a footer contains the text: 'webXaccelerator is © 2010 by Global Marine Networks LLC. All Rights Reserved. [view license]'.



Detailed information regarding administration of the RedPort wXa Router is documented in the **RedPort wXa Administrator Guide**.

5 Connecting to VoIP Web Administrator

The VoIP Web Administrator is a front-end editor for the XML Configuration for FreeSwitch upon which the RedPort VoIP system is based. For advanced FreeSwitch configuration information beyond the scope of this Administrator Guide, see [Appendix A – References](#).

The VoIP Web Administrator can be accessed in different ways dependent upon your model:

1. For wXa 400 series: Open a web browser and enter URL **http://<IP address of wXa LAN >** to connect to the **VoIP Web Administrator**. For example, when using the default LAN IP address the URL will be **http://192.168.10.1**.
2. For RedPort wXa 503 model only: Open a web browser to the wXa Router Web Administrator detailed in [Connecting to wXa Router Web Administrator](#).



Then select **Services > VOIP PBX**

OR

Open a web browser and enter URL **http://IP address of wXa LAN >:8080**

Note: IP address 192.168.10.1 is the default IP address for the wXa LAN port. 8080 is the port number for the VoIP system on the wXa-503.

5.1 Logins

5.1.1 superadmin

The **superadmin** login should be used for configuration of the VoIP system. This login displays the full functionality and configuration of your system.

```
Login: superadmin  
Password: webxaccess
```



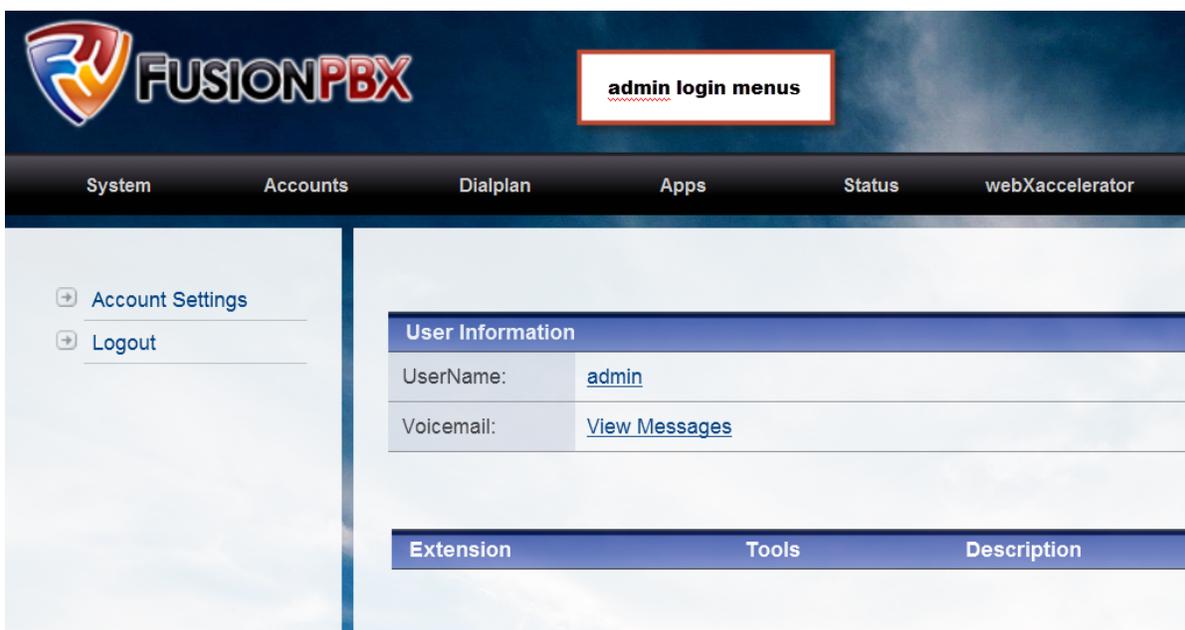
The screenshot shows the FusionPBX interface with the 'superadmin login menus' highlighted in a red box. The navigation bar includes System, Accounts, Dialplan, Apps, Status, Advanced, and webXaccelerator. The left sidebar contains Account Settings, Logout, Modules, Settings, Variables, Activation, and Database Manager. The main content area displays 'User Information' with fields for Username (superadmin) and Voicemail (View Messages). Below this is a table header with columns for Extension, Tools, and Description.

5.1.2 admin

The **admin** login can be used to limit certain administrative rights. This login will allow access for configuration of accounts but will not allow access to **Modules, Settings, Variables, Activation** and the **Database Manager** menus.

Login: admin

Password: webxaccess



The screenshot shows the FusionPBX interface with the 'admin login menus' highlighted in a red box. The navigation bar includes System, Accounts, Dialplan, Apps, Status, and webXaccelerator. The left sidebar contains Account Settings and Logout. The main content area displays 'User Information' with fields for Username (admin) and Voicemail (View Messages). Below this is a table header with columns for Extension, Tools, and Description.

6 Starting your VoIP System

During initial setup of the VoIP system, you will need to perform several steps before you can make a call on your VoIP system. This chapter gives a chronological view of these steps which include registering with your provider, activating your system, configuring your Analog phones (for applicable systems with analog ports), saving your configuration and restarting the system, checking the system status and verifying the caller id's for your lines.

6.1 Register with your Provider

1. Locate the MAC address on the sticker attached to the back of the unit. The MAC address is located beneath the bar code and will have a format similar to this example:

`a098050102be`

2. Record your MAC address here:

3. Contact your provider for your license key and voice plan information.

Your provider will work with you to recommend a voice plan for your business and then will provide you with a license key similar to this example:

`U2FsdGVkX18UVgWXtZtVg+pmBwGVG8bVL9qbX2Hws9niu9dyzd5C+R2yrYkXO7h9f
uSOBhesnoT6 uC4CcKWDNQ++`

4. Record your license key here: _____

5. Record your DID (Direct Inward Dialing) number here:

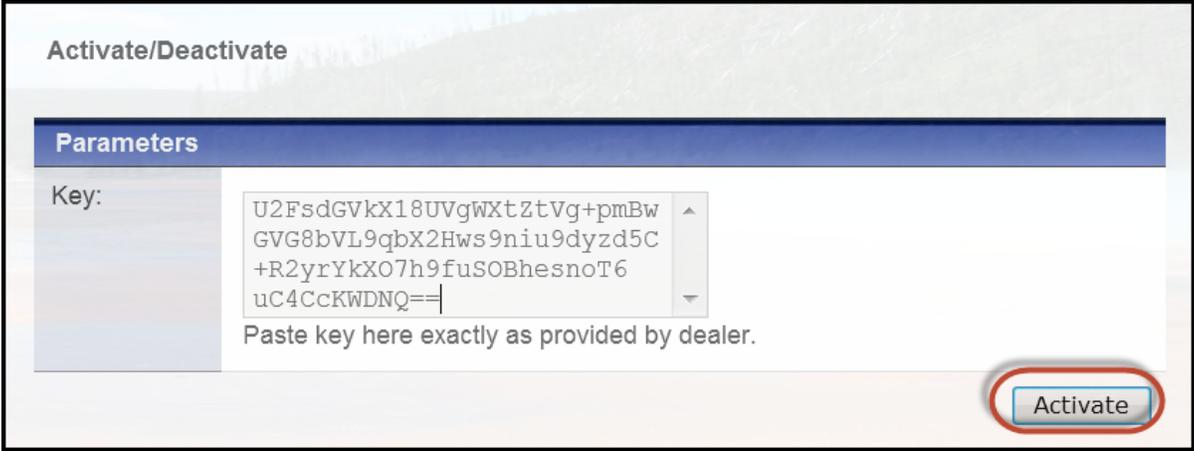
6. Record your UID (VoIP System ID Number or User ID) here:

For more information on how DID and UID numbers are used, see [Caller IDs for Crew Lines and Post-Paid Lines](#) and [Adding an Extension](#).

6.2 Activating your System

Follow these steps to activate your VoIP system:

1. Login to the **VoIP Web Administrator** - See [Connecting to VoIP Web Administrator](#)
2. Click on the **System** menu
3. Click on the **Activation** link on the left
4. Paste in the license key given to you by your provider in the **Key** field



Activate/Deactivate

Parameters

Key:

```
U2FsdGVkX18UVgWxtZtVg+pmBw
GVG8bVL9qbX2Hws9niu9dyzd5C
+R2yrYkXO7h9fuSOBhesnoT6
uC4CcKWDNQ==
```

Paste key here exactly as provided by dealer.

Activate

5. Click the **Activate** button
6. Click **Save** at the bottom of the screen. This saves the configuration and restarts the PBX system. If you select the **Restart PBX** button at the top of the screen, it will restart the PBX system but will not save changes to the configuration. See [Saving your Configuration and Restarting PBX](#).

Note: The unit must have access to an internet connection for the activation to work. Activations cannot be performed unless the unit can access the remote RedPort servers. This internet connection can be through a broadband satellite system such as VSAT or FleetBroadband. The connection can also be through a land based internet connection.

Warning: Be sure to click the button at the bottom of the screen.

6.3 Configuring your Analog Phones

Analog phones are connected into the Analog ports on your unit. Depending upon your unit, you may have a maximum of 8 analog ports.

Model	Analog Ports
wXa-400	No Analog Ports
wXa-404	A1-A4
wXa-408	A1-A8
wXa-503	A1-A4

Each of your analog phone lines will need to be configured in the **VoIP Web Administrator** before use. Before configuring the analog phones though you will need to determine if the analog phone is intended to be used as a **Post-Paid Line** or a **Crew Line**.

Post-Paid Lines operate like a typical phone line in a home or office. A Post-Paid Line configured phone is connected to the system, placed in a certain location, and anyone can make calls on the line assuming they have access to the location. Calls are paid for during a regular billing cycle after calls have been made hence the name “Post-Paid”.

Crew Lines (or Pre-Paid Lines) are set up to allow phone access to anyone that has a valid pincode or number associated with an account that has been prepaid for a certain amount of phone time, typically in minutes.

For example, suppose you want all crew members to be able to make calls from the phone in the lounge. If you set up the lounge phone as a post-paid line, everyone that has access to the lounge will be able to make calls and the bill for the phone line will go to your ship’s office. If you set up your lounge phone as a Crew Line, everyone that has access to the lounge will be able to make calls only if they have purchased a Pre-Paid amount of phone minutes. When they make a call, they will be prompted for their pincode. Their total phone usage on their Pincode is recorded and deducted from their balance of minutes. They do not receive a bill because the phone usage time was “Pre-Paid”.

To configure an analog line, follow these steps:

1. Login to the **VoIP Web Administrator**
2. Go to **System > Activation**
3. Scroll down to see the **Analog Lines** available for your system (a partial screenshot is below)

Analog Lines	
A1	
Name:	<input type="text" value="17135684983"/> Enter free text used to identify this line
Caller ID:	<input type="text" value="17135684983"/> Must be a valid assigned phone number
Crew Line:	<input type="text" value="false"/> Select true to enable pincode dialing on this line
A2	
Name:	<input type="text" value="17135684983"/> Enter free text used to identify this line
Caller ID:	<input type="text" value="17135684983"/> Must be a valid assigned phone number
Crew Line:	<input type="text" value="false"/> Select true to enable pincode dialing on this line

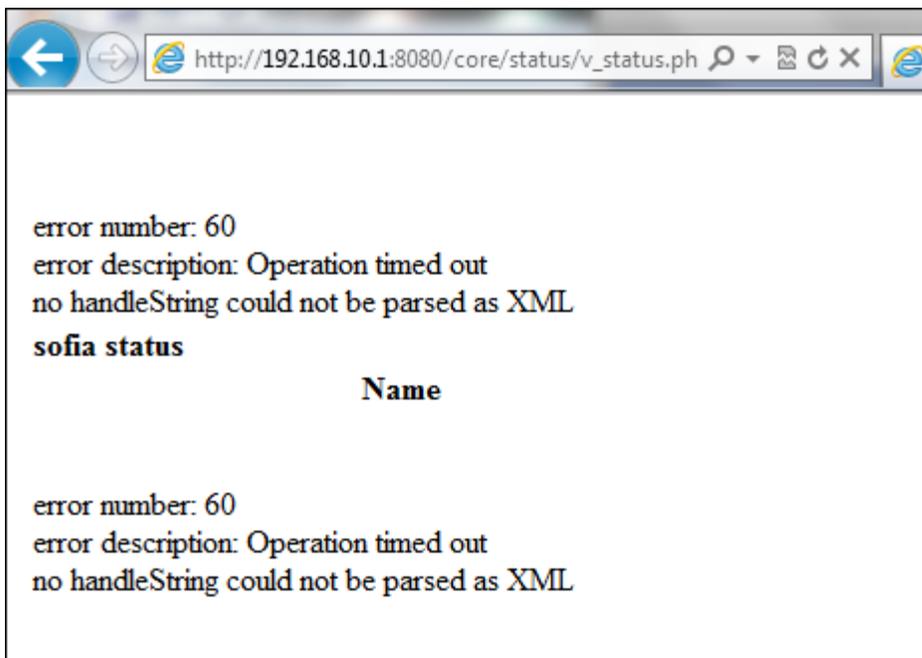
4. Physically connect an analog phone to an analog port
5. Locate the configuration fields for the analog port that was connected (example: A1)
6. Enter a **<name>** for the line, e.g. *Lounge* or *Captain's Office*
7. The Caller ID field should be set by default to your phone number at this time. If you have multiple phone numbers assigned to your ship and wish to use a different number for this line, you can enter the appropriate number here. You can also use system variables for fields such as this one to make editing easier in the future. See [Using System Variables](#). After the configuration is saved, note that the value in this field may change dependent upon your Crew Line value chosen below. See [Caller IDs for Crew Lines and Post-Paid Lines](#).
8. Select the **Crew Line** value: **false** = Post-Paid Line; **true** = Crew Line (Pre-Paid)
9. Repeat steps 4-8 for additional analog port configurations
10. Click **Save** at the bottom of the page. See [Saving your Configuration and Restarting PBX](#).

6.4 Saving your Configuration and Restarting PBX

In order to complete the configuration process you must click the **Save** button. This saves the configuration for your phones and restarts the PBX system.

Warning: There is also a **Restart PBX** button at the top of the screen. This will restart the PBX but will not save any changes to your configuration.

A PBX restart will take roughly 30 to 60 seconds. If you click on **PBX Status** and get an error message similar to the screenshot below this means the restart process has not completed.



Wait a few seconds and then click the refresh button on your browser to see the PBX Status.



6.5 Checking the System Status

You will want to perform a couple of system checks to verify your system is ready to go. These are detailed below.

6.5.1 Decoder Status

Verify that your VoIP system is registered with the provider and that it is authorized to make phone calls:

1. Login to the **VoIP Web Administrator**
2. Go to **System > Activation**
3. Verify **DECODER STATUS** has a HOST IP address
4. Verify the **Status** field displays **OK**

SN	Host IP	Netmask	Port	Status	Delay
21251	209.160.77.225	255.255.255.255	4999	OK	47ms

Note: The DECODER STATUS must have a HOST IP address and the Status field must display OK. This verifies that the VoIP system is registered with the provider, the account is active and the system is *allowed* to make phone calls.

6.5.2 PBX Status

Verify that your PBX system is ready to make calls using the following steps:

1. Login to the **VoIP Web Administrator**
2. Go to **System > Activation**
3. Click **PBX Status** to view the state of the PBX
4. Verify that the **Vobal** table line has a state of **REGED**

Name	Type	Data	State
internal	profile	sip.mod_sofia@192.168.1.100:5060	RUNNING (0)
external	profile	sip.mod_sofia@192.168.1.100:5080	RUNNING (0)
internal-ipv6	profile	sip.mod_sofia@[::1]:5060	RUNNING (0)
Vobal	gateway	sip:90001@192.168.77.2	REGED
192.168.1.100	alias	internal	ALIASED

Note: A state of REGED indicates that the system is active, the PBX is registered with the VoIP system and it is *ready* to make calls.

6.6 Caller IDs for Crew Lines and Post-Paid Lines

Below is an example of the configuration of two analog lines. A1 was configured as a Crew Line (Pre-Paid). A2 was configured as a Post-Paid line. Note the values for the Caller ID fields which are updated after the configuration is saved and the PBX is restarted.

Note: Setting the crew line to **true** will force the caller id to be the UID for the system. Any value entered into the caller ID field will be ignored. On refreshing the screen after **Save** you will note that the correct UID is entered in the caller ID field. Setting the crew line to **false** followed by **Save** will revert the caller ID to the primary DID for the unit. You will need to change the DID to a secondary value if you wish to assign an alternate DID for a particular line.

The Caller ID for Crew Lines (Pre-Paid) will be the VoIP System ID number (UID).

The Caller ID for Post-Paid Lines will be the DID telephone number (or extension).

Analog Lines	
A1	A1: Example of Crew Line (Pre-Paid)
Name:	Lounge Enter free text used to identify this line
Caller ID:	21251 Must be a valid assigned phone number
Crew Line:	true Select true to enable pincode dialing on this line
A2	A2: Example of Post-Paid Line
Name:	Captain's office Enter free text used to identify this line
Caller ID:	17135684983 Must be a valid assigned phone number
Crew Line:	false Select true to enable pincode dialing on this line

7 Making a Call

7.1.1 Post-Paid Lines

To dial out on a **Post-Paid** line use the following dialing sequence:

<COUNTRY_CODE> <PHONE_NUMBER>

Examples for Post-Paid Lines:

To call the GMN office in the US, dial *1 865 379 8723*

To call a cell phone in Argentina (country code 54), dial *54 9 290 160 2064*

7.1.2 Crew Lines (Pre-Paid)

To dial out on a **Crew** line (**Pre-Paid**) first acquire a 10 digit prepaid pincode from your provider.

For these examples, we will use a pincode of 1234567890.

Dial out on a Crew Line using the following sequence:

<COUNTRY_CODE> <PHONE_NUMBER> * <PINCODE>

OR

<COUNTRY_CODE><PHONE_NUMBER><WAIT FOR VOICE PROMPT><PINCODE>

Examples for Crew Lines (Pre-Paid):

To call the GMN office in the US, dial *1 865 379 8723 *1234567890*

To call a cell phone number in Argentina (country code 54),

dial *54 9 290 160 2064 *1234567890*

7.1.3 Calling a Vessel

To call the vessel, dial the phone number provided to you by your dealer. The VoIP PBX Auto



Attendant will answer and play a pre-recorded message. See [The IVR \(Interactive Voice Response\) Menu](#) for how to customize the pre-recorded message.

To connect to one of the analog extensions, dial 101 x where x is port 1-8; e.g. line A1 is extension 1011, line A2 is extension 1012, etc.

8 Configuring Extensions

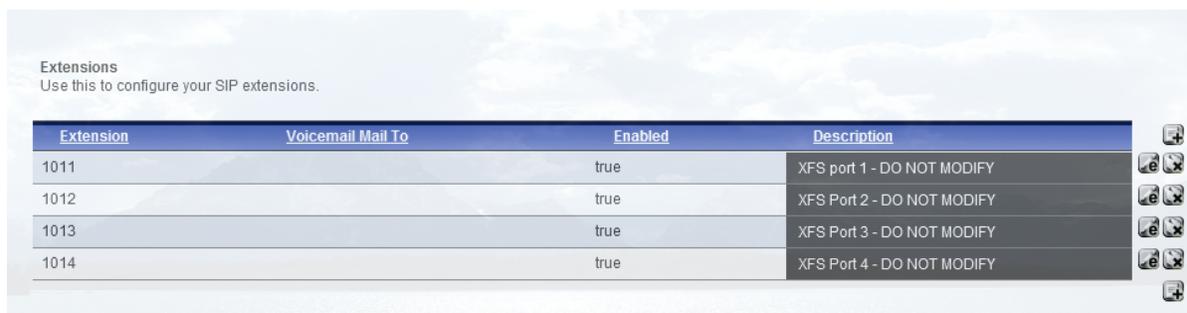
The RedPort VoIP System features an unlimited number of extensions that can be added to the system. There are certain extensions that are used for the Analog ports that you will need to be aware of and these are detailed in this chapter. In addition to the analog extensions, additional extensions can be added for software phones and SIP (Internet) Phones. Examples of the configuration of some of the most popular software phones and SIP Phones are included in the chapter. See [Configuring a SNOM Phone](#) and [Configuring xLite Software Phone](#).

8.1 Analog Extensions

A number of extensions are predefined in the VoIP system for the Analog ports. Analog extensions begin with 101 followed by a number to indicate the analog port; e.g. 1012 is the extension for analog port 2.

To view the current extensions:

1. Login to the **VoIP Web Administrator**
2. Go to **Accounts > Extensions**



The screenshot shows the 'Extensions' page in the VoIP Web Administrator. It includes a header with the title 'Extensions' and a sub-header 'Use this to configure your SIP extensions.' Below this is a table with four columns: 'Extension', 'Voicemail Mail To', 'Enabled', and 'Description'. The table lists four extensions: 1011, 1012, 1013, and 1014. Each row has a 'true' value in the 'Enabled' column and a description for each XFS port. To the right of each row are icons for edit, delete, and add.

Extension	Voicemail Mail To	Enabled	Description
1011		true	XFS port 1 - DO NOT MODIFY
1012		true	XFS Port 2 - DO NOT MODIFY
1013		true	XFS Port 3 - DO NOT MODIFY
1014		true	XFS Port 4 - DO NOT MODIFY

Note: Your system may have 0-8 analog extensions listed depending upon your particular VoIP model.

8.2 Adding an Extension

To add a new extension:

1. Login to the **VoIP Web Administrator**
2. Go to **Accounts > Extensions**

Extensions
Use this to configure your SIP extensions.

Extension	Voicemail Mail To	Enabled	Description
1011		true	XFS port 1 - DO NOT MODIFY
1012		true	XFS Port 2 - DO NOT MODIFY
1013		true	XFS Port 3 - DO NOT MODIFY
1014		true	XFS Port 4 - DO NOT MODIFY

3. Click  at the bottom right of the list of defined extensions to add a new extension
4. Enter values for the fields detailed below (this is the minimum configuration needed to create a new extension):

Extension: The new extension number (e.g. 2000)

Extension Add

Extension:

Enter the extension here. The default configuration allows 2 - 7 digit extensions.

Effective Caller ID Name: The Caller ID Name used internally

Effective Caller ID Number: The Caller ID Number used internally

In this example when a call is placed from this phone to another phone on the same internal system, the caller ID name that will be displayed is *Captain* for extension *2000*.

Effective Caller ID Name:

Enter the effective caller id name here.

Effective Caller ID Number:

Enter the effective caller id number here.

Outbound Caller ID Name: Enter the VoIP System ID (UID) for a Crew Line (Pre-Paid) or the DID number for a Post-Paid Line

Outbound Caller ID Number: Enter the VoIP System ID (UID) for a Crew Line (Pre-Paid) or the DID number for a Post-Paid Line

Note: Your DID and UID numbers are included with your VoIP Activation Key from your provider. See [Register with your Provider](#) for more information.

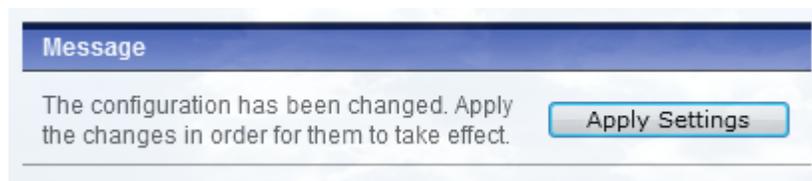
In this example, extension 2000 is going to be a Post-Paid Line set up for the captain. The DID number is used for the Outbound Caller ID Name and the Outbound Caller ID Number.

Outbound Caller ID Name:	<input type="text" value="17135684983"/> Enter the outbound caller id name here.
Outbound Caller ID Number:	<input type="text" value="17135684983"/> Enter the outbound caller id number here.

Warning: The Outbound Caller ID Name and Number must be set to the proper values (either DID or UID) corresponding to the type of line (Crew Line or Post-Paid). If the values are not set correctly, the extension will not be valid.

Enabled: Verify that this field is set to **true**

5. Click **Save**
6. Click **Apply Settings** in the **Message** box at the top of the screen



The set of configuration options above reflects the minimum configuration needed for a new extension. More than likely, you will want to assign a user to a particular extension in order to take advantage of the voice mail capabilities. To do so, the user account will need to be created and then the user is added to the configuration of the extension. For additional information, refer to [User Accounts](#).

8.3 Using System Variables

When defining extensions, it is useful to set a system variable for values that will be used frequently and for values that are subject to change. Many system variables are pre-defined and can be used easily in your configuration. For example, if you use a system variable for the DID, you can use the system variable name in fields that require the DID phone number. If your DID number changes in the future then you would only need to set the value for the system variable instead of modifying all of the extensions where the DID number is hardcoded.

To access system variables:

1. Login to the **VoIP Web Administrator**
2. Go to **System > Variables**
3. Note the predefined **Default** variables for **outbound_caller_id** and **outbound_caller_name**

Variables
Define preprocessor variables here.

Codecs

Name	Value	Enabled	Description
global_codec_prefs	G722,PCMU,PCMA,GSM	true	
outbound_codec_prefs	G722,PCMU,PCMA,GSM	true	

Defaults

Name	Value	Enabled	Description
outbound_caller_id	MODIFY_WITH_CALLER_ID	true	Default caller id number for SIP extensions. This must be a valid (DID - i.e. phone number) or decoder serial number. SIP extensions can override this value.
outbound_caller_name	MODIFY_WITH_CALLER_NAME	true	Default value in free text which identifies SIP extensions. SIP extensions can override this value.
call_debug	false	true	

Variables can be modified by clicking on the  at the end of the variable line and changing the appropriate fields.

For example, to edit the **outbound_caller_id** with our current DID, we would change the Value to reflect our DID number.

Variable Edit

Name:
Enter the variable name here.

Value:
Enter the variable value here.

We would edit the **outbound_caller_name** similarly as in this example:

Variable Edit

Name:
Enter the variable name here.

Value:
Enter the variable value here.

Remember to click **Save** to save your configuration.

After the variables are set, the variables can be referenced in the extensions as `${{outbound_caller_name}}` and `${{outbound_caller_id}}`. The screenshot below provides an example of using these variables when editing an extension:

Outbound Caller ID Name:	<input type="text" value="\${{outbound_caller_id}}"/> Enter the outbound caller id name here.
Outbound Caller ID Number:	<input type="text" value="\${{outbound_caller_name}}"/> Enter the outbound caller id number here.

8.4 Configuring a SNOM Phone

Example: We want to configure a SNOM 300 phone as extension 2000 for the Captain's office.

For SNOM phone configuration, follow these steps:

1. Connect the SNOM phone to your VoIP network via Ethernet cable
2. Use the arrow keys on the SNOM phone to scroll through the SNOM menu until you find **Information > IPAdr**
3. Click  on the phone to select the menu item
4. Note the IP Address of the phone
5. Open a web browser using the Phone IP address
6. On the SNOM configuration page, select **Identity 1** on the left menu
7. On the **Login** tab, enter values for **Display Name, Account, Password, Registrar, Outbound Proxy**, and **Authentication Username**

The **Outbound Proxy** is set to the IP address of the primary interface for the device.

For wXa-40x: Outbound Proxy = IP address of the Ethernet interface

For wXa-503: Outbound Proxy = WAN IP address

It is highly recommended that these interfaces be assigned static IP addresses so that the IP phones will not stop working when a DHCP configured unit is assigned a different IP address after reboot.

[Login](#) [SIP](#) [NAT](#) [RTP](#)

Login Information:

Identity active: on off ?

Displayname: ?

Account: ?

Password: ?

Registrar: ?

Outbound Proxy: ?

Failover Identity: ?

Authentication Username: ?

Mailbox: ?

Ringtone: ?

Custom Melody URL: ?

Display text for idle screen: ?

Ring After Delay (sec): ?

Record Missed Calls: on off ?

Record Dialed Calls: on off ?

Record Received Calls: on off ?

8. Click **Apply**
9. Click **Save**
10. On the **RTP** tab, turn off **RTP Encryption**

[Login](#) [SIP](#) [NAT](#) [RTP](#)

RTP Identity Settings:

Codec 1:	G.711u	?
Codec 2:	G.711a	?
Codec 3:	G.722	?
Codec 4:	G.726-32	?
Codec 5:	GSM FR	?
Codec 6:	G.729AB	?
Codec 7:	G.723.1	?
Packet Size:	20 ms	?
Full SDP Answer:	<input checked="" type="radio"/> on <input type="radio"/> off	?
Symmetrical RTP:	<input type="radio"/> on <input checked="" type="radio"/> off	?
RTP Encryption:	<input type="radio"/> on <input checked="" type="radio"/> off	?
Dynamic G.726 payload:	<input checked="" type="radio"/> on <input type="radio"/> off	?
G.726 Byte Order:	<input checked="" type="radio"/> RFC3551 <input type="radio"/> AAL2	?
SRTP Auth-tag:	<input checked="" type="radio"/> AES-32 <input type="radio"/> AES-80	?
RTP/SAVP:	off	?
Media Transport Offer:	UDP	?
Media Transport Offer Setup:	active	?

11. On the **Login** tab, click **re-register**
12. Your SNOM phone should now be registered and you can begin making test calls.

To reset or reboot a SNOM phone, use the down arrow key on the phone to get to the **Configuration** menu. Then use the right arrow key to scroll through menu selections for **Reg** (registration), **Reset**, or **Reboot**. Press the enter (or ) key to select the menu item.

8.5 Configuring an X-Lite Software Phone

X-Lite is a free software phone, i.e. softphone that can be downloaded at <http://www.counterpath.com>. There is an X-Lite version for Mac or Windows systems. Download and install X-Lite on a computer system that is on the same LAN as your VoIP system. Also available from CounterPath are Bria softphones for IOS (iPhone and iPad) and Android systems. There is a charge for the Bria edition. These softphones allow users to

configure their mobile phones so that they make and receive calls over their GMN VoIP Satellite links. For more information, see <http://www.counterpath.com>.

For X-Lite configuration, follow these steps:

1. Open the X-Lite application
2. Select **Softphone > Account Settings**
3. Under User Details enter values for **Account Name, User ID, Domain, Password, Display Name** and **Authentication Name**
4. Under **Domain Proxy**, check **Proxy** and enter the **Proxy Address**

The **Proxy Address** is set to the IP address of the primary interface for the device.

For wXa-40x: Outbound Proxy = IP address of the Ethernet interface

For wXa-503: Outbound Proxy = WAN IP address

It is highly recommended that these interfaces be assigned static IP addresses so that the IP phones will not stop working when a DHCP configured unit is assigned a different IP address after reboot.

SIP Account

Account | Voicemail | Topology | Presence | Transport | Advanced

Account name: xLite

Protocol: SIP

Allow this account for

Call

IM / Presence

User Details

* User ID: 1002

* Domain: gmn-usa.com

Password: ●●●●

Display name: user2

Authorization name: 1002

Domain Proxy

Register with domain and receive calls

Send outbound via:

Domain

Proxy Address: 192.168.1.105

Dial plan: #1\|a.T;match=1;prestrip=2;

OK Cancel

5. Click **OK**
6. Your **X-Lite** phone will connect with the VoIP system and you will be able to make calls from the softphone through your system

9 User Accounts

User accounts need to be created in order to assign a user to a particular extension and to use voicemail on the VoIP system. A user can login to the user account and check their call records for their extension and listen to their voice mail. The two steps needed for this configuration are detailed in this chapter: **Creating a User Account**, and **Assigning a User Account to an Extension**.

9.1 Creating a User Account

User accounts are needed to allow voice mail to be bound to a user ([Assigning a User Account to an Extension](#)) and to also allow the user to view call records made on that extension. When a user logs into the **VoIP Web Administrator** using his username and password he will be able to hear any pending voice mail directly from the browser. He will also be able to see the inbound/outbound calls made on his extension.

To create a user account:

1. Login to the **VoIP Web Administrator**
2. Go to **Accounts > User Manager**
3. Click  at the lower right of the user manager table
4. Enter the user information in the form
5. Click **Create Account** when finished

To sign up as a member, please fill out this form completely. All fields are required.

Username:	<input type="text" value="1012"/>
Password:	<input type="password" value="...."/>
Confirm Password:	<input type="password" value="...."/>
First Name:	<input type="text" value="Captain"/>
Last Name:	<input type="text" value="Gary"/>
Email:	<input type="text" value="captain@gmail.com"/>

Note: The user name and password created is used to login to the **VoIP Web Administrator** system for voice mail access. In this example Captain Gary would login with username 1012 to access his voicemail.

9.2 Assigning a User Account to an Extension

Assign a User Account an existing extension using the following steps:

1. Login to the **VoIP Web Administrator**
2. Go to **Accounts > Extensions**
3. Locate the extension in the table and click  to edit the extension
4. Use the drop-down menu for **User List** to select the user to assign to the extension
5. Enter or edit the **Voicemail Password** if desired

Extension Edit

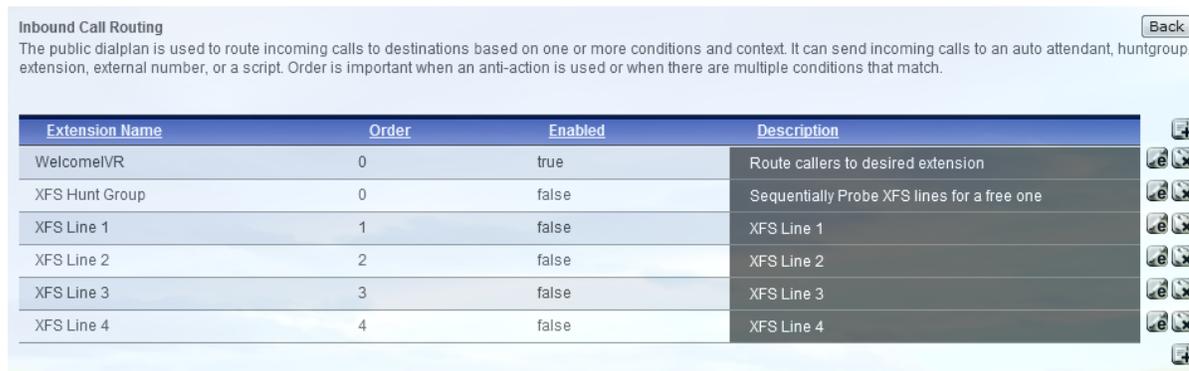
Extension:	<input type="text" value="1012"/> Enter the extension here. The default configuration allows 2 - 7 digit extensions.
Password:	<input type="password" value="••••"/> Enter the password here. Password: 1012
User List:	<input type="text" value="1012"/>  sign users to this extension. <ul style="list-style-type: none">10121101SusanadminIsolterosuperadmin the user list and it will be cre:
Voicemail Password:	<input type="password" value="••••"/> Enter the voicemail password here. Password: 1012

10 Routing Incoming Calls to Different Destinations

The **Dialplan Manager** handles the routing of calls for your VoIP system. The system is configured by default to route your incoming calls to the **WelcomeIVR**, the Interactive Voice Response system. It is important to understand where and how the **Dialplan Manager** is configured to route incoming calls in case there is a need to change this routing in the future.

To view the **Inbound Call Routing** Table follow these steps:

1. Login to the **VoIP Web Administrator**
2. Go to **Dialplan Manager > Inbound Routes**



The screenshot shows the 'Inbound Call Routing' configuration page. At the top, there is a 'Back' button and a descriptive paragraph: 'The public dialplan is used to route incoming calls to destinations based on one or more conditions and context. It can send incoming calls to an auto attendant, huntgroup, extension, external number, or a script. Order is important when an anti-action is used or when there are multiple conditions that match.' Below this is a table with the following data:

Extension Name	Order	Enabled	Description
WelcomeIVR	0	true	Route callers to desired extension
XFS Hunt Group	0	false	Sequentially Probe XFS lines for a free one
XFS Line 1	1	false	XFS Line 1
XFS Line 2	2	false	XFS Line 2
XFS Line 3	3	false	XFS Line 3
XFS Line 4	4	false	XFS Line 4

Welcome IVR

Note that the first entry is the **WelcomeIVR**. This Dialplan is enabled by default. If you were to look at the configuration for the **WelcomeIVR** (by clicking the edit button), you would find the following **Conditions and Actions** have been defined. These conditions and actions are written in a programming language that is understood by the PBX system. For example, the highlighted line shows the coding that is needed to program the system to answer all calls coming in. This language is fully documented in the FreeSwitch Reference manual listed in [Reference documents](#) of this guide. If you need to modify the default rules for the inbound calls on your system, please refer to the reference documents for detailed information on the programming language syntax and rules.

Conditions and Actions
The following conditions, actions and anti-actions are used in the public to direct call flow. Each is processed in order until you reach the action tag which tells the system what action to perform. You are not limited to only one condition or action tag for a given extension.

Tag	Type	Data	Order
condition	context	public	0
condition	destination_number	^(.*)\$	1
action	set	ignore_early_media=true	2
action	answer		3
action	sleep	2000	4
action	ivr	WelcomeWXA	10

Additional information about the IVR can be found in [Setting up the IVR \(Interactive Voice Response\)](#).

XFS Hunt Group

The second entry in the Inbound Route table is an example of an XFS Hunt Group. This is disabled by default. A Hunt Group can be defined to sequentially ring a group of numbers (for example your analog lines). Say for example the Hunt Group is defined to have 2 lines in this order - 1011, 1012. The system will try 1011 and if it is busy, it will ring 1012.

XFS Line X

The third entry in the Inbound Route table is an example of an XFS Line definition. This is disabled by default. Individual lines can be set up to ring for a specific call. For example, each Analog port can be assigned a unique DID or phone number. When a call is made to that specific US number (DIDs for other countries are also available) the analog extension bound to that number (and only that extension) will ring. Note that the system comes with one DID which is routed by the dialing plan. Additional fees apply for secondary DIDs. The system can have as many inbound DIDs as the user wants.

The VoIP Web Administrator is a front-end editor for the XML Configuration for FreeSwitch upon which the RedPort VoIP system is based. For advanced FreeSwitch configuration information beyond the scope of this Administrator Guide, see [Appendix A – References](#).

11 Call Detail Records (CDRs)

Detailed information about calls for your VoIP system is kept in the Call Detail Records (CDR). The records contain useful call details such as the source and destination of the call and the duration. The records can be filtered, for example, to find all calls originating from a certain extension. The data can also be downloaded as a CSV file for use in spreadsheet applications or your billing system for Post-Paid calls.

11.1 Accessing the CDR

To access the Call Detail Records:

1. Login to the **VoIP Web Administrator**
2. Go to **Apps**
3. The **Call Details Records** Table will display

Call Detail Records

Call Detail Records (CDRs) are detailed information on the calls. The information contains source, destination, duration, and other useful call details. Use the fields to filter the information for the specific call records that are desired. Then view the calls in the list or download them as comma separated file by using the 'csv' button.

Start: Source: Duration:
 Status: Destination: Bill:

Start	CID Name	Source	Destination	Duration	Bill	Status	csv
2012-05-16 13:55:45	21251	Lounge	18656907670	11	9	normal_clearing	>
2012-05-16 13:55:16	21251	Lounge	101	1	1	normal_clearing	>
2012-05-15 17:17:19	21251	Lounge	*10102	64	64	normal_clearing	>
2012-05-15 17:02:39	17135684983	Captain™s office	101	1	1	normal_clearing	>
2012-05-15 17:02:25	17135684983	Captain™s office	4000	1	1	normal_clearing	>
2012-05-15 16:39:38	21251	Lounge	*10102	81	81	normal_clearing	>
2012-05-15 15:48:23	17135684983	Captain™s office	101#	0	0	no_route_destination	>

- Start:** Start time of call
- CID Name:** Caller ID. See [Caller IDs for Crew Lines and Post-Paid Lines](#)
- Source:** Identifier for where the call originated
- Destination:** Lists the number which was dialed
- Duration:** Time of call (in seconds)

Bill: Time of call for billing (in seconds). Note the Bill time is sometimes less than Duration time because of call setup time.

Status: Status information for if the call completed

11.2 Detailed Call Information

Additional detail about a specific call can be accessed by clicking the > button on the call record.

Start	CID Name	Source	Destination	Duration	Bill	Status	
2012-05-16 13:55:45	21251	Lounge	18656907670	11	9	normal_clearing	 

The example is below.

Call Detail Records: Details

CID Name:	21251
CID Number:	Lounge
Destination:	18656907670
Context:	default
Start:	2012-05-16 13:55:45
Answer:	2012-05-16 13:55:47
End:	2012-05-16 13:55:56
Duration:	11
Bill Seconds:	9
Hangup Cause:	NORMAL_CLEARING
UUID:	1a40835c-809f-e111-b277-a098050102be
Bleg UUID:	6932855c-809f-e111-b277-a098050102be

11.3 Filtering the Call Detail Records

To filter the Call Detail Records, use the fields at the top of the CDR Table.

For example, to filter all calls that originate from the Lounge, enter *Lounge* in the filter block and then click **Filter**. The records for this example appear below.

Call Detail Records

Call Detail Records (CDRs) are detailed information on the calls. The information contains source, destination, duration, and other useful call details. Use the fields to filter the information for the specific call records that are desired. Then view the calls in the list or download them as comma separated file by using the 'csv' button.

Start: Source: Duration:
Status: Destination: Bill:

Start	CID Name	Source	Destination	Duration	Bill	Status	csv
2012-05-16 13:55:45	21251	Lounge	18656907670	11	9	normal_clearing	>
2012-05-16 13:55:16	21251	Lounge	101	1	1	normal_clearing	>
2012-05-15 17:17:19	21251	Lounge	*10102	64	64	normal_clearing	>
2012-05-15 16:39:38	21251	Lounge	*10102	81	81	normal_clearing	>
2012-05-15 15:45:40	21251	Lounge	*10102	75	75	normal_clearing	>
2012-05-15 15:30:27	21251	Lounge	*10102	5	5	normal_clearing	>
2012-05-14 13:20:29	21251	Lounge	186559991950	11	11	normal_clearing	>

11.4 Saving the Records to a CSV File

To save the CDR records to a CSV file, click **CSV**:

Call Detail Records

Call Detail Records (CDRs) are detailed information on the calls. The information contains source, destination, duration, and other useful call details. Use the fields to filter the information for the specific call records that are desired. Then view the calls in the list or download them as comma separated file by using the 'csv' button.

Start: Source: Duration:
Status: Destination: Bill:

Start	CID Name	Source	Destination	Duration	Bill	Status	csv
2012-05-16 13:55:45	21251	Lounge	18656907670	11	9	normal_clearing	>
2012-05-16 13:55:16	21251	Lounge	101	1	1	normal_clearing	>
2012-05-15 17:17:19	21251	Lounge	*10102	64	64	normal_clearing	>
2012-05-15 16:39:38	21251	Lounge	*10102	81	81	normal_clearing	>
2012-05-15 15:45:40	21251	Lounge	*10102	75	75	normal_clearing	>
2012-05-15 15:30:27	21251	Lounge	*10102	5	5	normal_clearing	>
2012-05-14 13:20:29	21251	Lounge	186559991950	11	11	normal_clearing	>

The CSV file can be saved for processing by your billing system or opened in a spreadsheet:

	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O
1	cdr_id	v_id	caller_id	caller_id	destination	context	start_stamp	answer_stamp	end_stamp	duration	billsec	hangup_c	uuid	bleg_uuid	accountco
2		48	1	21251	Lounge	1.87E+11	default	5/14/2012 13:20	5/14/2012 13:20	5/14/2012 13:20	11	11	NORMAL_51e7581a-bc285f1a-e99d-e111-f		
3		58	1	21251	Lounge	*10102	default	5/15/2012 15:30	5/15/2012 15:30	5/15/2012 15:30	5	5	NORMAL_2c3f936c-c49e-e111-b277-a098-f		
4		59	1	21251	Lounge	*10102	default	5/15/2012 15:45	5/15/2012 15:45	5/15/2012 15:46	75	75	NORMAL_80881b8d-c69e-e111-b277-a098-f		
5		61	1	21251	Lounge	*10102	default	5/15/2012 16:39	5/15/2012 16:39	5/15/2012 16:40	81	81	NORMAL_f706e016-ce9e-e111-b277-a098-f		
6		64	1	21251	Lounge	*10102	default	5/15/2012 17:17	5/15/2012 17:17	5/15/2012 17:18	64	64	NORMAL_61abbd5a-d39e-e111-b277-a098-f		
7		65	1	21251	Lounge	101	default	5/16/2012 13:55	5/16/2012 13:55	5/16/2012 13:55	1	1	NORMAL_d1cb3e4b-809f-e111-b277-a098-f		
8		66	1	21251	Lounge	1.87E+10	default	5/16/2012 13:55	5/16/2012 13:55	5/16/2012 13:55	11	9	NORMAL_1a40835c-6932855c-809f-e111-f		
9															
10															
11															
12															

12 The IVR (Interactive Voice Response) Menu

The IVR Menu displays the IVR (Interactive Voice Response) that is configured to specify what happens when a number gets called in the VoIP system. This component of your VoIP system allows messages to be played and then processes any interactive responses in order to direct calls to particular extensions or phone lines. By default, the system is configured with one IVR which is the WelcomeWXA IVR. There can be any number of IVRs in your system to define different processes if needed. For example, your ship may have two different phone numbers with a different IVR Menu set up for each one. After the IVR Menu is defined, the Inbound Routes menu would then be used to specify which phone numbers should be routed to which IVR. See [Routing Incoming Calls to Different Destinations](#) for information on configuring the Inbound Routes.

The IVR configuration controls many features of the interactive voice response including specifying the recordings that your callers hear when the phone is answered. It is also the configuration that allows the administrator to specify if it is acceptable for a caller to enter an extension number for dialing.

To access the IVR Menu:

1. Login to the **VoIP Web Administrator**
2. Go to **Apps> IVR Menu**
3. The **IVR Menu List** is displayed – WelcomeWXA is the default IVR

IVR Menu List

The IVR Menu plays a recording or a pre-defined phrase that presents the caller with options to choose from. Each option has a corresponding destination. The destinations can be extensions, voicemail, IVR menus, hunt groups, FAX extensions, and more.

Name	Extension	Direct Dial	Enabled	Description
WelcomeWXA	5000	true	true	Welcome to webXaccelerator

To edit (or copy) the WelcomeWXA IVR:

1. Login to the **VoIP Web Administrator**
2. Go to **Apps> IVR Menu**
3. Click  on the WelcomeWXA line

4. The **IVR Menu Edit** screen will open
5. Use the **Copy** button at the top if you want to make a copy of WelcomeWXA for your IVR

IVR Menu Edit

The IVR Menu plays a recording or a pre-defined phrase that presents the caller with options to choose from. Each option has a corresponding destination. The destinations can be extensions, voicemail, IVR menus, hunt groups, FAX extensions, and more.

Name:	<input type="text" value="WelcomeWXA"/> Enter a name for the IVR menu.
Extension:	<input type="text" value="5000"/> Enter the extension number.
Greet Long:	<input type="text" value="ivr/ivr-enter_destination_telephone_number.wav"/> The long greeting is played when entering the menu.
Greet Short:	<input type="text" value="ivr/ivr-enter_ext.wav"/> The short greeting is played when returning to the menu.
Timeout:	<input type="text" value="10000"/> The number of milliseconds to wait after playing the greeting or the confirm macro.
Direct Dial:	<input type="text" value="enabled"/> Define whether callers can dial directly to extensions and feature codes.
Show Advanced	<input type="button" value="Advanced"/>
Enabled:	<input type="text" value="enabled"/> Define whether the IVR Menu is enabled or disabled.
Description:	<input type="text" value="Welcome to webXaccelerator"/> Enter a description.

Fields of interest

Greet Long: This is the greeting that is played when a caller calls your VoIP system. Use the pull-down menu to change your system greeting, if desired. There are numerous selections of existing greetings in the menu from which to choose.

Greet Short: This is the greeting that is played when a caller enters invalid information or no information. It is generally a shorter greeting than Greet Long, e.g. without the typical introduction. Use the pull-down menu to change your greeting, if desired.

For information on how to create and upload your own greetings for **Greet Long** and **Greet Short**, see [Changing your Recordings](#).

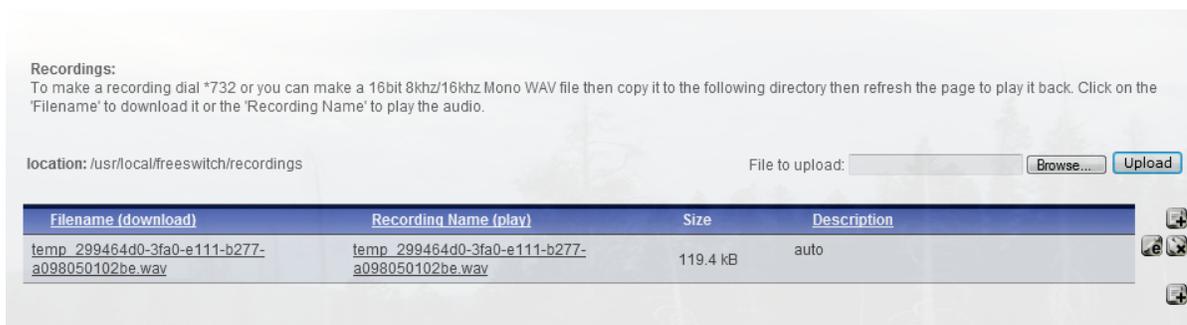
Direct Dial: Enable this option if you want to allow callers to enter extension numbers instead of having to pick from a voice menu.

For additional IVR Menu information, refer to the FreeSWITCH documents listed in [Appendix A – References](#).

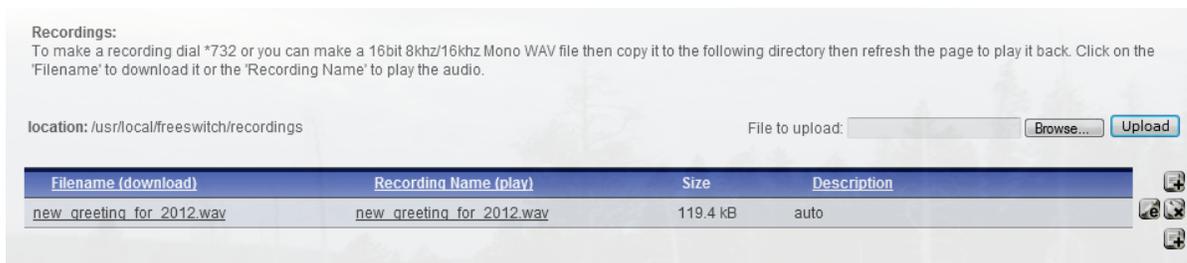
13 Changing your Recordings

Make a new recording to be used in your IVR Menu with these steps:

1. Login to the **VoIP Web Administrator**
2. Go to **Apps > Recordings**
3. Dial ***732** to create a new recording
4. Refresh your browser page to see your new recording file



5. Click on the **Recording Name (play)** entry to play the new audio file
6. Click  to edit the Filename and Recording Name
7. Click **Save** - The **Recordings** table will display your new filename



Note: Any external software application that generates a wave file (.wav) can be used to create a greeting for an IVR. Simply create the wave file and then follow the steps for [Uploading a New Recording](#).

Upload a New Recording:

1. Login to the **VoIP Web Administrator**
2. Go to **Apps > Recordings**
3. Click **Browse...** to locate your **.wav** file

File to upload:

4. Click **Upload** to upload it to the VoIP appropriate directory

After your new greeting file is created or uploaded, you will be able choose your new recordings from your IVR drop-down edit menus for **Greet Long** and **Greet Short**.

```
misc
  phrase:
  say:
recordings
  new_greeting_for_2012.wav
sounds
  time/yesterday.wav
  time/day-5.wav
```

See [The IVR \(Interactive Voice Response\) Menu](#) for complete instructions on changing your greetings.

14 Using the Database Manager

The Call Detail Records are kept in a VoIP system database and are accessible for reporting as documented in [Call Detail Records \(CDRs\)](#). It is understandable that the VoIP system will generate many CDR database records which may need to be cleared out periodically. The frequency of clearing the database records will be different for every system so it is left up to the Administrator to determine the appropriate interval for his or her specific system.

Warning: Exercise extreme caution when using the Database Manager as any other actions other than those documented here can have adverse effects on your database

To clear out the database records, follow this procedure:

(Note this procedure is for **Advanced Users Only**)

1. Login to the **VoIP Web Administrator**
2. Go to **System > Database Manager**
3. The **FusionPBX** database is displayed

The screenshot shows the phpLiteAdmin v1.8.3 interface for the FusionPBX database. On the left, there is a list of tables under the 'FusionPBX' column. On the right, the 'Structure' tab is active, displaying a table with columns for 'Table', 'Action', and 'Records'. The table lists various tables and their corresponding actions and record counts.

Table	Action	Records
v_auto_attendant	Browse Structure SQL Search Insert Export Import Rename Empty Drop	0
v_auto_attendant_options	Browse Structure SQL Search Insert Export Import Rename Empty Drop	0
v_call_broadcast	Browse Structure SQL Search Insert Export Import Rename Empty Drop	0
v_call_center_agent	Browse Structure SQL Search Insert Export Import Rename Empty Drop	0
v_call_center_queue	Browse Structure SQL Search Insert Export Import Rename Empty Drop	0
v_call_center_tier	Browse Structure SQL Search Insert Export Import Rename Empty Drop	0
v_cdr	Browse Structure SQL Search Insert Export Import Rename Empty Drop	71
v_dialplan_includes	Browse Structure SQL Search Insert Export Import Rename Empty Drop	8
v_dialplan_includes_details	Browse Structure SQL Search Insert Export Import Rename Empty Drop	29
v_extensions	Browse Structure SQL Search Insert Export Import Rename Empty Drop	7
v_fax	Browse Structure SQL Search Insert Export Import Rename Empty Drop	0
v_gateways	Browse Structure SQL Search Insert Export Import Rename Empty Drop	1
v_group_members	Browse Structure SQL Search Insert Export Import Rename Empty Drop	13
v_groups	Browse Structure SQL Search Insert Export Import Rename Empty Drop	5
v_ivr_menu	Browse Structure SQL Search Insert Export Import Rename Empty Drop	4
v_menu	Browse Structure SQL Search Insert Export Import Rename Empty Drop	0

4. From the list of tables in the **FusionPbx** column on the left, select table **v_cdr**

FusionPBX

- v_auto_attendant
- v_auto_attendant_options
- v_call_broadcast
- v_call_center_agent
- v_call_center_queue
- v_call_center_tier
- v_cdr**
- v_dialplan_includes
- v_dialplan_includes_details

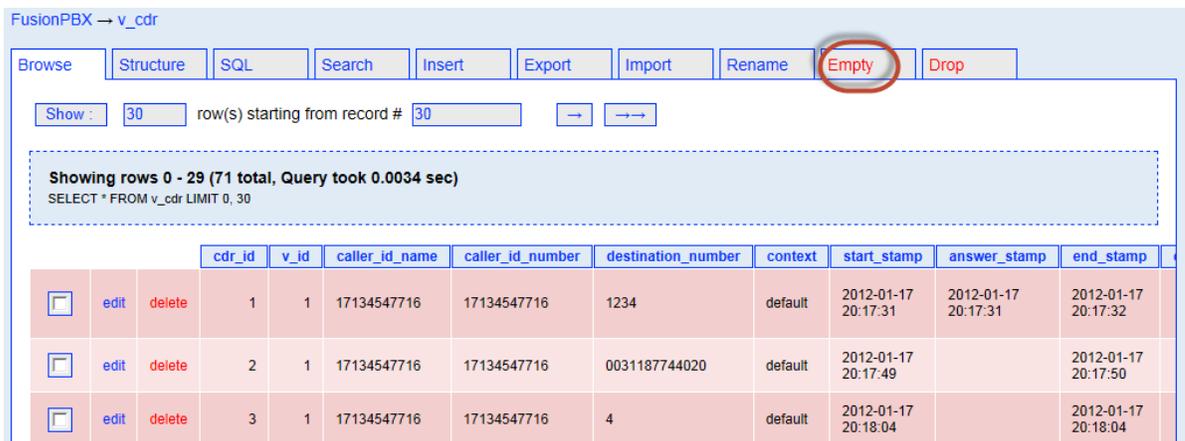
5. The v_cdr table is displayed:



The screenshot shows the FusionPBX interface for the v_cdr table. At the top, there are navigation buttons: Browse, Structure, SQL, Search, Insert, Export, Import, Rename, Empty, and Drop. Below these buttons, it says "Show: 30 row(s) starting from record # 30". A message box indicates "Showing rows 0 - 29 (71 total, Query took 0.0034 sec)" with the SQL query "SELECT * FROM v_cdr LIMIT 0, 30". Below the message is a table with the following columns: cdr_id, v_id, caller_id_name, caller_id_number, destination_number, context, start_stamp, answer_stamp, and end_stamp. The table contains three rows of data.

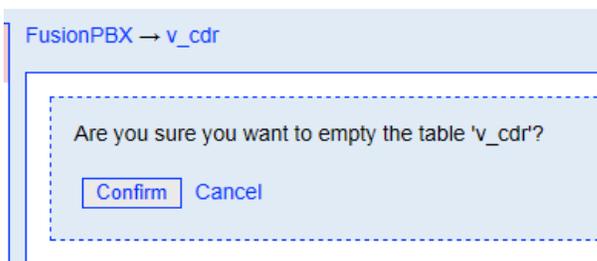
			cdr_id	v_id	caller_id_name	caller_id_number	destination_number	context	start_stamp	answer_stamp	end_stamp
<input type="checkbox"/>	edit	delete	1	1	17134547716	17134547716	1234	default	2012-01-17 20:17:31	2012-01-17 20:17:31	2012-01-17 20:17:32
<input type="checkbox"/>	edit	delete	2	1	17134547716	17134547716	0031187744020	default	2012-01-17 20:17:49		2012-01-17 20:17:50
<input type="checkbox"/>	edit	delete	3	1	17134547716	17134547716	4	default	2012-01-17 20:18:04		2012-01-17 20:18:04

6. Click **Empty** to clear out the database table entries for the v_cdr table



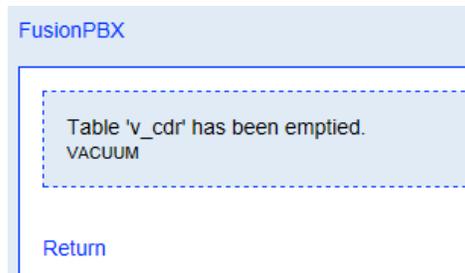
The screenshot shows the same FusionPBX interface as before, but the "Empty" button in the top navigation bar is circled in red.

7. Click **Confirm** on the pop-up window



The screenshot shows a confirmation dialog box with the text "Are you sure you want to empty the table 'v_cdr'?" and two buttons: "Confirm" and "Cancel".

8. The following message is displayed

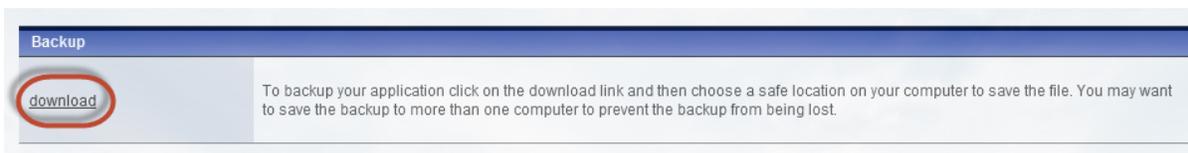


9. Click **Return** to return to the **Database Manager**

15 Backing up your System

To backup your system:

1. Login to the **VoIP Web Administrator**
2. Go to **Status > System Status**
3. Scroll down to bottom of page to the **Backup** section
4. Click **download**



5. Download the backup database file (file name **fusionpbx.db**, by default) to a safe location on your computer or network

Appendix A - References

Global Marine Networks, LLC. (2012). *RedPort wXa Administrator Guide*.

Minessale, A., Collins, M. S., & Schreiber, A. (2010). *FreeSwitch 1.0.6*. Birmingham, B27 6PA, UK.: PACKT Publishing.

Minessale, A., Schreiber, D., Collins, M. S., & Chandler, R. (2012). *FreeSWITCH Cookbook*. Birmingham B3 2PB, UK.: PACKT Publishing.

OSTAG (Open Source Telephony Advancement Group). (n.d.). Retrieved from FreeSWITCH wiki: <http://wiki.freeswitch.org>

Quick Start Guide for VoIP

Congratulations on the purchase of your RedPort VoIP appliance. With RedPort VOIP you will increase satellite data transfer efficiency and decrease airtime costs for your voice calls. The webXaccelerator provides up to 8 concurrent calls which can be placed on either pincode metered crew lines or open lines. Calls on average consume about 6kbps which means that you can enjoy up to 25 minutes of talk time per MByte of data consumed.

The following information will get you up and running quickly. Please refer to the wXa Administrator Guide (on the DVD included with your purchase) for additional information.

Port Definitions for WXA-5XX series

ETH1 is configured as WAN with DHCP and should be connected to your primary satellite unit. View wXa Quick Start guide or manual for instructions on configuring your wXa network interfaces.

Port Definitions for WXA-4XX series

ETH2 is configured as the LAN port enabled for DHCP. Connect this port to your LAN.

PBX Administrative Page

Use a web browser and URL `http://IP_ADDRESS_OF_WXA:8080` to access the VoIP PBX

```
user: superadmin
```

```
password: webxaccess
```

Activation Procedure

1. Locate the wXa MAC address on the sticker attached to the back of the unit and record it.
2. Contact your provider, decide on a voice plan, and provide him with the MAC address of your unit.
3. Your provider will then provide you with a key which will look something like the following `U2FsdGVkX1/6air+xaNc65sMx6mqoOjgfxDcakOjyddQ+A3508MakbohbvZOByKzQN4rzEEimsuS BesDZq9zBA==` along with your units phone number and VOIP identification number.
4. Login to the PBX by using a web browser and connecting to port 8080 on the IP address of the webXaccelerator.
5. Select the System menu and then select the Activation link on the left.
6. Paste the key provided by your dealer into the appropriate form entry and then push the Activate button.
7. Next review the analog line assignment and change the names of the lines to be more descriptive (e.g. crew, captain, lounge, etc).
8. Enable crew lines by setting the appropriate entries to "true".
9. Hit save to complete the activation. Note it takes about 30 seconds for the activation to complete. Do not navigate away from the form while the activation is being processed.

Status

Check the status of your VoIP activation before making the first call. Under System->Activation confirm that the decoder status is OK. The status display should look similar to the following:

SN	Host IP	Netmask	Port	Status	Delay
21254	209.160.77.225	255.255.255.255	4999	OK	44ms

Under System->Activation push the "PBX Status" button to view the state of the PBX. It should look like the following:

Name	Type	Data	State
internal	profile	sip:mod_sofia@192.168.0.3:5060	RUNNING (0)
external	profile	sip:mod_sofia@192.168.0.3:5080	RUNNING (0)
internal-ipv6	profile	sip:mod_sofia@[::1]:5060	RUNNING (0)
Vobal	gateway	sip:90001@192.168.77.2	REGED
192.168.0.3	alias	internal	ALIASED

Phones

Plug in analog POTS phones into the desired A1-A4 ports on the front of the unit.

Dialing Instructions

To dial out on a standard line use the following dialing sequence:

COUNTRY_CODE (followed by) PHONE_NUMBER

For example to call the GMN office in the US you would dial *18653798723*. To dial a cell phone in Argentina (country code 54) you would dial *5492901602064*.

To dial out on a crew line first acquire 10 digit prepaid pincodes from your provider. Then dial using the following sequence:

COUNTRY_CODE (followed by) PHONE_NUMBER * PINCODE

For example to call the GMN office in the US using a crew line you would dial something like *18653798723*1234567890*. A cell phone number in Argentina (country code 54) would be dialed as *5492901602064*1234567890* where the 1234567890 corresponds to a sample pincode.

To call the vessel dial the phone number provided to you by your dealer. The VoIP PBX auto attendant will answer and play a pre-recorded message. To connect to one of the analog extensions dial 101X where X corresponds to port 1-4. e.g. line A1 is extension 1011.

Boot/Halt

To power on a WXA-5XX router follow the instructions in the corresponding manual or Quick

Quick Start Guide for wXa

Congratulations on the purchase of your webXaccelerator appliance. With webXaccelerator you will increase satellite data transfer efficiency and decrease airtime costs while preventing unwanted costly transmissions. We are confident that webXaccelerator will serve you well.

The following information will get you up and running quickly. Please refer to the webXaccelerator user guide (on the DVD included with your purchase) for additional information.

Port Definitions

ETH0 is configured as the LAN port with IP address 192.168.10.1.

ETH1 is configured as WAN with DHCP and should be connected to your primary satellite unit.

ETH2 is configured as WAN2 with DHCP and should be connected to your backup satellite unit.

WLAN is configured as the WiFi port (on units with WiFi) with IP address 192.168.20.1.

WiFi Hotspot

Protocol: 802.11g (a and b are disabled by default)

Security: WPA2 (WPA and WEP disabled by default)

SSID: webXaccelerator

Secret: webXaccelerator

WiFi IP: 192.168.20.1

Router Administration Web Page

IP: 192.168.10.1 - Use a web browser and URL <http://192.168.10.1> on the LAN port or via WiFi.

user: admin

password: webxaccess

Configuring WAN and WAN2 Ports

Both the primary (WAN) and backup (WAN2) ports are configured with DHCP by default. Use the following to configure for use with static IP addressing.

1. Login to the router administration web page
2. Go to **Interfaces->WAN**
3. Set the Type to **Static** (currently DHCP)
4. Enter the IP address for the WAN port and the Gateway then click **Save**.
5. Go to **System->General** and enter the DNS servers in the appropriate boxes
6. Uncheck **Allow DNS server list to be overridden by DHCP/PPP on WAN** and click **Save**

Web Compression

Web compression is disabled by default. The following steps are required to enable compression.

1. Contact your supplier for a webXaccelerator account username and password.
2. Login to the router administration web page
3. Go to **Services->Proxy server**
4. Click on **Upstream Proxy**, enable it and enter the information below:

Hostname: xweb.gmn-usa.com

Username: Assigned by your supplier

Password: Assigned by your supplier
Compression level: More compression accelerates the connection
sacrificing picture quality

5. Test the compression by viewing an internet website

Captive Portal

The captive portal is disabled by default. The following steps are required to enable it.

1. Login to the router administration web page
2. Go to **Firewall->Traffic shaper** and enable it
3. Go to **Services->Captive portal** and enable it

You will now be taken to a login page every time you try to access the internet with a web browser. Access to the internet will be restricted unless a valid pincode is entered. Contact your supplier for pincodes.

Captive Portal WiFi Considerations

Once the captive portal is enabled, access to the web admin at <http://192.168.10.1> is lost unless you enter a pincode. The following enables access to the router administration web page without having to enter a pincode.

1. Login to the router administration web page
2. Go to **Services->Captive portal**
3. Click on **Allowed IP addresses**
4. Click on **+** to add a **To** rule for 192.168.10.1, then select **Save**

Firewall Considerations

By default, both WAN and WAN2 are configured to be "Open" allowing unrestricted traffic originating on the internet. LAN and WLAN (the WiFi interface) are also configured "Open" allowing all traffic to pass through the router onto the internet. Access rules for all network interfaces should be modified to meet the operational environment. Firewall rules are defined on the router administration web page under **Firewall->Rules**. Click on the tab for the interface that you want to configure. Then add/remove rules to suit your specific environment.

Console

The supplied NULL modem cable can be used to gain console access through the serial port using settings 9600 8N1.

Backup/Restore

On the router administration web page, select **Diagnostics->Backup/Restore** to save a copy of your router's settings.

Boot/Halt

The webXaccelerator uses an internal solid state silicon disk drive (SSD) to store its configuration and cache webpages. Care should be taken when shutting down the unit to prevent loss of data and lengthy disk checks on startup. To properly shutdown the unit, login to the web admin and select **Diagnostics->Halt**. Wait one full minute before removing power.

In the Box

The following items are included in every webXaccelerator shipment:

- ⤴ Base unit. Either rack or wall mount base unit
- ⤴ WiFi antenna for WiFi enabled units
- ⤴ DB-9 NULL modem cable
- ⤴ US or EU compatible external power supply
- ⤴ Resource DVD
- ⤴ This Quick Start guide.

Appendix B – Glossary of Terms

Term	Meaning
DID	Direct Inward Dialing number
IVR	Interactive Voice Response
MAC Address	Media Access Control Address; Unique identifier for a network interface
PBX	Private Branch Exchange; Telephone exchange system that makes connections between internal telephones of a private organization
SIP	Session Initiation Protocol
SIP Phone	Uses the Session Initiation Protocol (SIP) to provide the traditional call functions of a telephone
UID	VoIP System ID Number or User ID
VoIP	Voice over Internet Protocol
wXa	Global Marine Networks RedPort webXaccelerator